MG3000-R32

WEB User Guide

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1 General Introduction

MG3000-R32 series products are developed as a type of advanced VoIP access equipment according to the features of broadband construction and to the development demand of voice business.

Besides the functions of the traditional 4-port/8-port gateway, MG3000-R32 series products improve their reliability by the intelligent routing technology. MG3000-R32 provides flexible access methods; it supports not only phone connecting but also SPC telephone exchange connecting and trunk cable connecting. With its built-in high-capability voice chip, it may be a substitute for small SPC exchange equipment to those small and medium-sized enterprise branches.

Figure 1–1 shows the basic configuration flow.

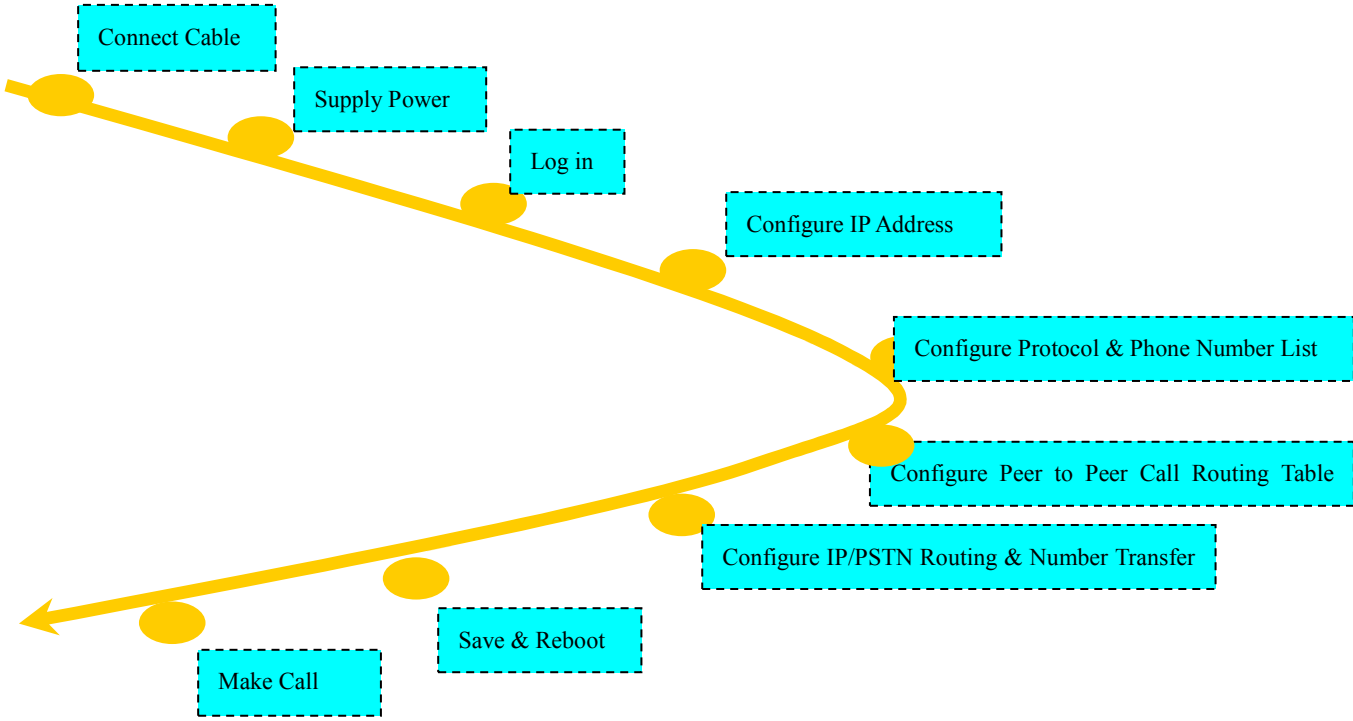


Figure 1-1 MG3000-R32 Quick Configuration Flow

2 Connect Cable & Supply Power

The quantity of FXO/FXS ports may differ in different models, and users shall follow the actual labels on the equipment. FXO port may be connected to PBX or an analog trunk cable; FXS port may be connected to FXO port or a telephone.

3 Initial Configuration

Users may perform WEB management by WAN port. Its IP address in initial configuration is 192.168.9.237。

System default user: admin (password: admin)

4 Log in the WEB Management Interface

Fist, perform a command of running ping WAN port IP address, such as “ping 192.168.9.237” to confirm the accessibility of WAN port; then, start up the browser in PC and input the IP address of WAN port in the address blank, such as”192.168.9.237”, and then press “enter” to back to the WEB login interface, as Figure4-1shows.

Figure 4-1 MG3000-R32 Login Interface

Input the user name and password in the login interface and click the “Enter” button. If successful, the user will enter into the Quick Setup Interface of WEB management, as Figure4-2 shows.

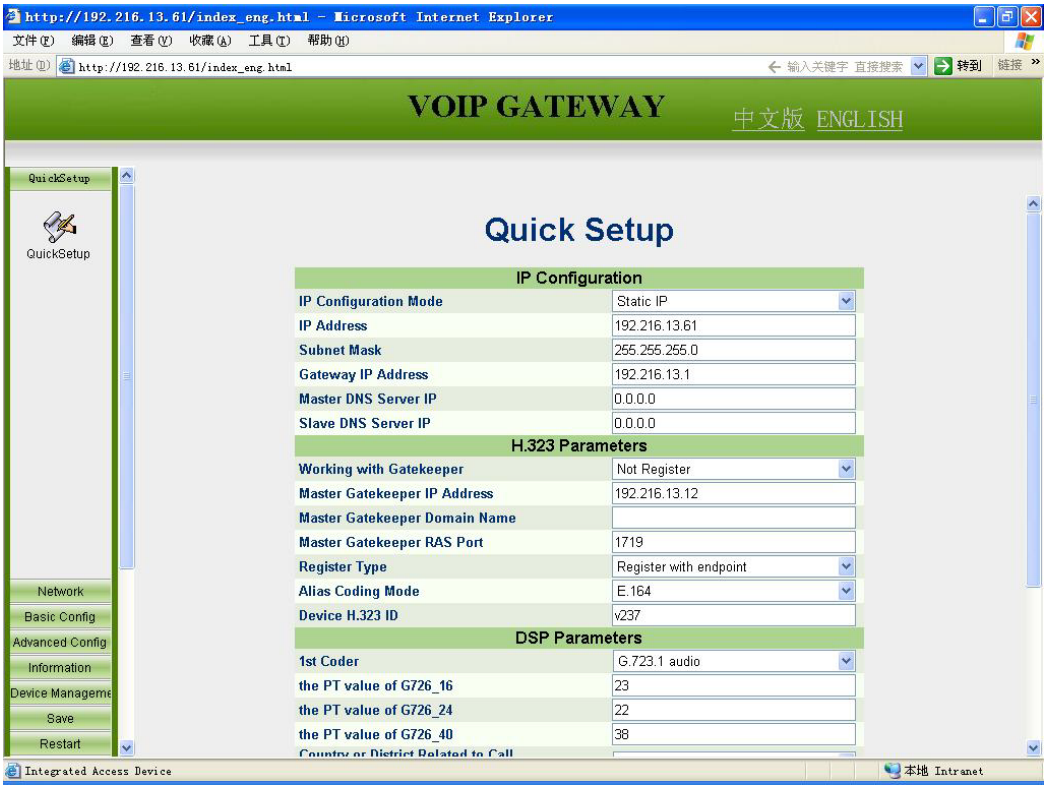


Figure 4-2 MG3000-R32 H323 Version Quick Setup Interface

5 Quick Setup

5.1 Setup Network Parameter

When entering into the management interface, the user needs to configure the IP address of the IAD, that is, to set up [IP Configuration] in the interface of [Quick Setup] as Figure4-2 shows.

- If the LAN of the IAD adopts dynamic address location, the user needs to choose to acquire IP address through DHCP.
- If the LAN of the IAD adopts static address, then the user needs to choose static IP and set up [IP Address], [Subnet Mask] and [Gateway IP Address].
- If the IAD adopts domain name registration method, the user needs to set up the IP address of the master/slave DNS server.

5.2 Setup Protocol Parameter

Before configuring the VoIP protocol parameter, the user needs to confirm the type of software version of MG3000-R32. For users with H.323 version, please refer to Chapter 5.2.1; for users with SIP version, please refer to Chapter 5.2.2.

5.2.1 Configuration of H.323 Protocol

Shown as Figure 4-2.

5.2.1.1 Peer to Peer Method

If it is peer to peer routing method, please choose [Not Register] in [Working with Gatekeeper] in the [Quick Setup] interface, and then directly refer to Chapter 5.3.

5.2.1.2 Registration Ways

5.2.1.2.1 IP Registration

- Choose [IP Register] in [Register to GK] in the interface of [Quick Setup]
- Fill in the IP address of GK in [Master Gatekeeper IP Address], such as "192.168.9.88".
- Configure the RAS port number of GK in [Master Gatekeeper RAS Port]. The default registered RAS port for H.323 Protocol is 1719.

5.2.1.2.2 Domain Name Registration

- Choose [Domain Name Register] in [Register to GK] in the interface of [Quick Setup].
- Fill in the domain name of GK in [Master Gatekeeper Domain Name], such as "gk@ghrd.com".
- Set up the IP address of the master/slave DNS server in the interface of [Network Setup].
- Configure the RAS port number of GK in [Master Gatekeeper RAS Port]. The default registered RAS port for H.323 Protocol is 1719.

5.2.1.2.3 Endpoint Registration

- Choose [Endpoint Register] in [Register Type] in [Quick Setup].
- Set up alias coding mode in [Alias Coding Mode] in [Quick Setup] as "E.164" or "E.164+Port H323 ID". If the user sets it up as "E.164+Port H323 ID", the user needs to set up H.323 ID of each port (allocated by the softswitch) in [Telephone Number] in [Quick Setup] at the same time.

5.2.1.2.4 Gatekeeper Registration

- Choose [Gatekeeper Register] in [Register Type] in [Quick Setup].

- Set up alias coding mode in [Alias Coding Mode] in [Quick Setup] as "E.164" or "E.164+Device H323 ID". If the user sets it up as "E.164+Device H323 ID", the user needs to set up H.323 ID (allocated by the softswitch) in [Device H.323 ID] in [Quick Setup] at the same time.

5.2.2 Configuration of SIP Protocol

Shown as Figure 5-1.

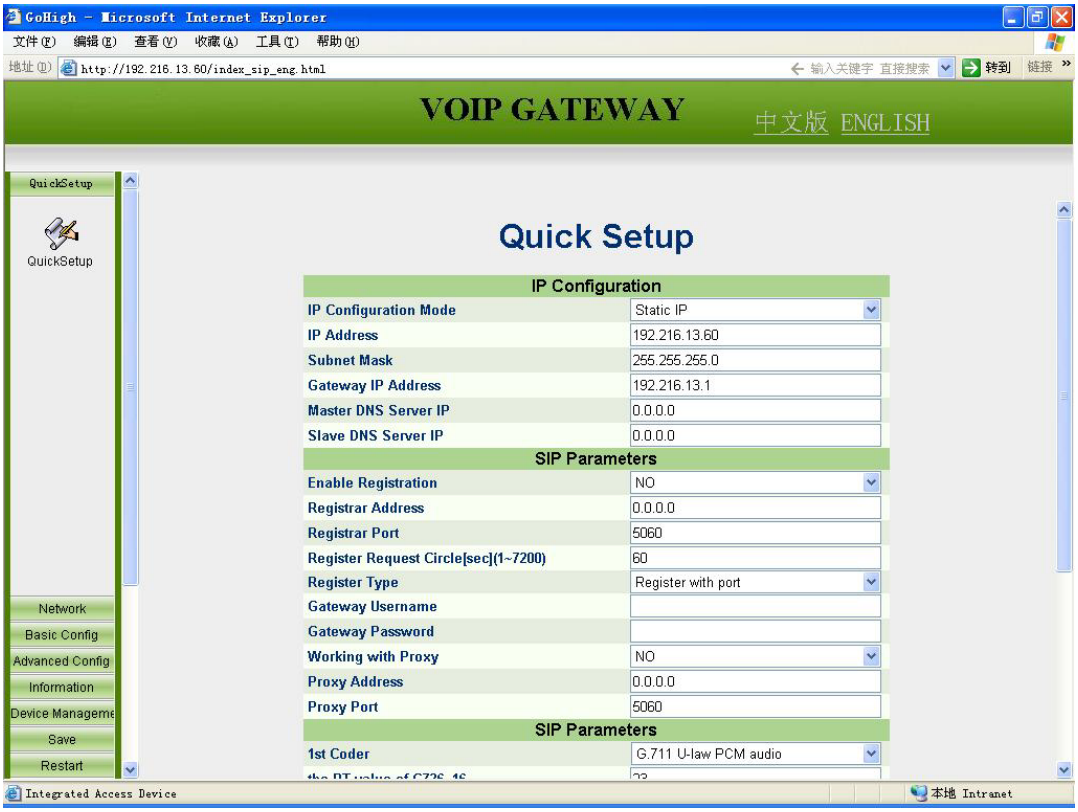


Figure 5-1 MG3000-R32 SIP Protocol Quick Setup Interface

5.2.2.1 Peer to Peer Method

If it is peer to peer routing method, please choose [No] in [Enable Registration] in the [Quick Setup] interface, and then directly refer to Chapter 5.3.

5.2.2.2 Registration Mode

5.2.2.2.1 IP Registration

- Choose [Yes] in [Enable Registration] in the [Quick Setup] interface.
- Fill in the IP address of SIP server in [Local Host Name].
- Fill in the IP address of SIP server in [Registrar Address], such as "192.168.9.88".
- Set up the signaling receiving port number of the registered server in

[Registrar Port]. The default registered port of SIP Protocol is 5060. Make the same setup in [Working with Proxy], [Proxy Address] and [Proxy Port].

- Set up the interval of time sending registration info in [Register Request Cycle].

5.2.2.2.2 Domain Name Registration

- Choose [Yes] in [Enable Registration] in [Quick Setup] interface.
- Fill in the domain name of the SIP server in [Local Host Name].
- Fill in the domain name of SIP server in [Registrar Address], such as "sip@ghrd.com".
- Set up the IP address of the master/slave DNS server in the interface of [Network Setup]
- Set up the signaling receiving port number of the registered server in [Registrar Port]. The default registered port of SIP Protocol is 5060. Make the same setup in [Working with Proxy], [Proxy Address] and [Proxy Port].
- Set up the interval of time sending registration info in [Register Request Cycle].

5.2.2.2.3 Endpoint Registration

- Choose "Register with Endpoint" in [Register Type].
- For each port, fill in the corresponding user name and password (allocated by the Softswitch) in [User Name]/[User Password] in [Telephone Number List] in [Quick Setup].

5.2.2.2.4 Gateway Registration

- Choose "Register with Gateway " in [Register Type].
- Fill in one user name and one user password, that is the gateway user name and password (allocated by the Softswitch) in [Gateway Username]/[Gateway Password] in [Quick Setup] interface.

5.3 Setup DSP Parameter

[DSP Parameter] in [Quick Setup] interface is designed for users to configure the coding mode. Because the standards of tone signals differ in different countries, users need to choose the

right place of application of the equipment so that the FXO port may detect the type of tone signals correctly.

- First Coder: preferred coding/decoding method of voice channels. Users may choose one from G.711 U-law PCM audio, G.723.1 audio, G.726_32 audio, G.711 A-law PCM audio, G.729 A audio, G.726_16 audio, G.726_24 audio and G.726_40 audio.
- PT value of G726_16 Coder: Payload type under G.726_16 Coder.
- PT value of G726_24 Coder: Payload type under G.726_24 Coder.
- PT value of G726_40 Coder: Payload type under G.726_40 Coder.
- Tone signal applicable countries or regions: choose the place of application of the equipment.

5.4 Setup the Endpoint Number List

Before configuring the Endpoint Number List, please confirm the software version of MG3000-R32 first. For users with H.323 version, please directly refer to Chapter 5.4.1; for users with SIP version, please directly refer to Chapter 5.4.2..

5.4.1 Setup the Endpoint Number List with H.323 version

Click the button of [Telephone Number List] in [Quick Setup] interface to set up the phone numbers, as Figure 5-2 shows.

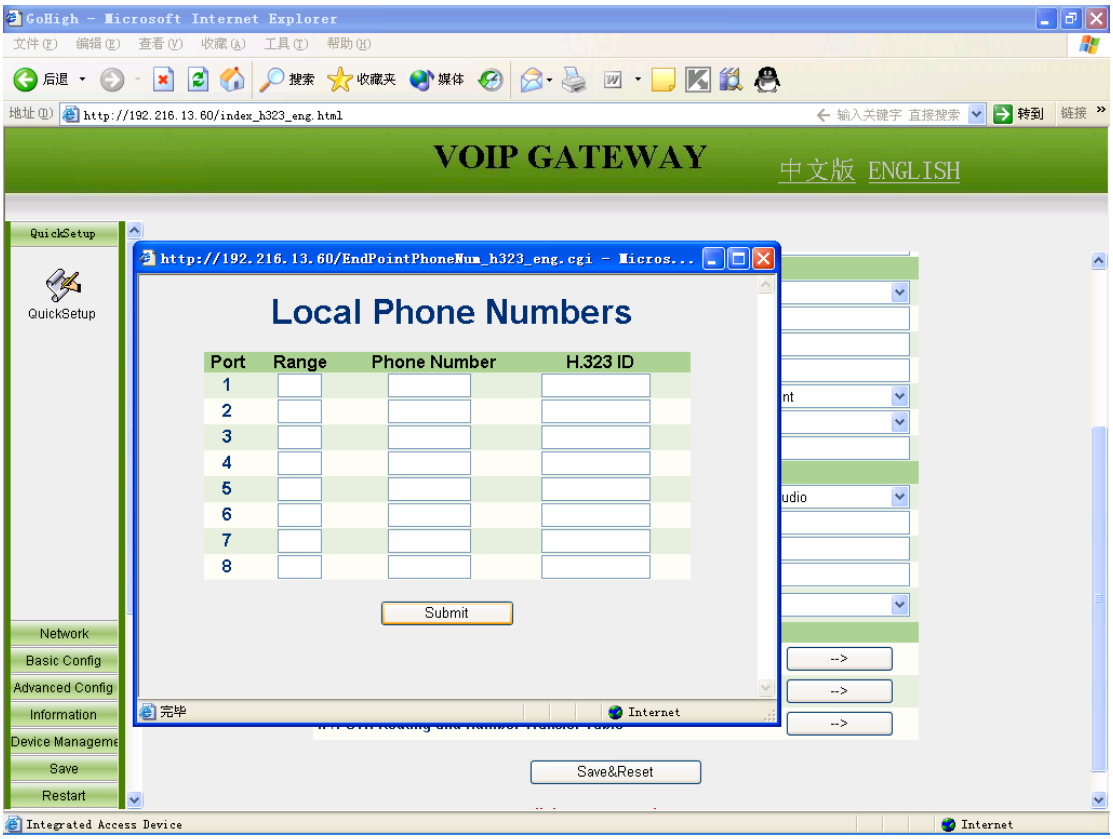


Figure 5-2 MG3000-R32 Phone Numbers Setup Interface with H323 Version

- Set up the phone number with each port.
- Meantime, set up H.323 ID for each port in [Local Phone Numbers] if the user chooses “E.164+Port H.323ID” in [Alias Coding Mode] in [Quick Setup].
- [Range] is designed for users’ convenience to input a group of continuous telephone numbers.

In the blank of [Range], the user may input a single port number, such as “2” and then input the telephone number and H.323 ID, thus meaning to setup the Port 2; the user may also input a range of ports, such as“ 1-5”, and then input the telephone number and H.323 ID; thus meaning the telephone numbers are increased in turn from Port 1 to Port 5, while the parameter of H.323 ID is shared among all the ports from Port 1 to Port 5

5.4.2 Setup the SIP Protocol Port Number List

To enter into the telephone number setup interface, just click the button of [Telephone Number List] in [Quick Setup], as shown in Figure 5-3：

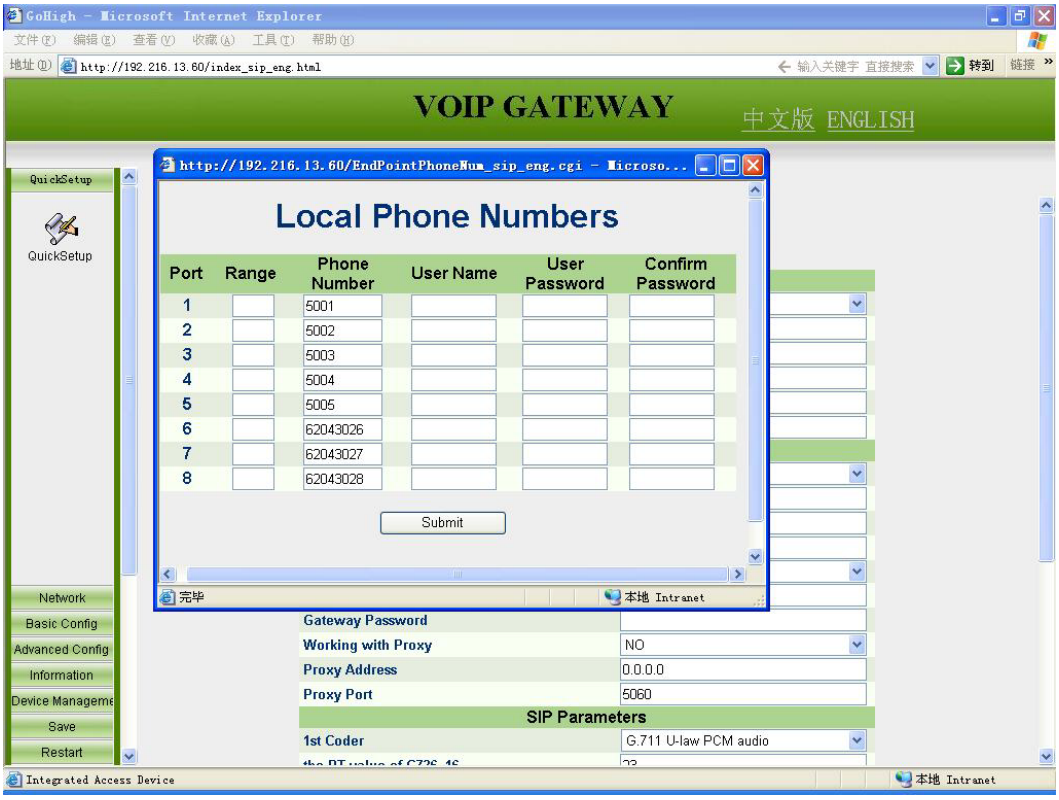


Figure 5-3 MG3000-R32 SIP Protocol Telephone Number Setup Interface

- Setup the telephone number of each port.
- The user needs to setup the user name and password for each port when an authorization from the Softswitch System is required in “Port Register Mode”.
- [Range] is designed for users’ convenience to input a group of continuous telephone numbers.

In the blank of [Range], the user may input a single port number, such as “2” and then input the telephone number, user name, user password and password confirmation, thus meaning to setup the Port 2; the user may also input a range of ports, such as “1-5”, and then input the telephone number, user name, user password and password confirmation; thus meaning the telephone numbers are increased in turn from Port 1 to Port 5, while the user name, user password and password confirmation are shared among all the ports from Port 1 to Port 5.

5.5 Peer to Peer Call Routing Table

To enter into the setup interface of peer to peer call routing table, click the button of [Peer to

Peer Call Routing Table] in [Quick Setup], as shown in Figure 5-4.

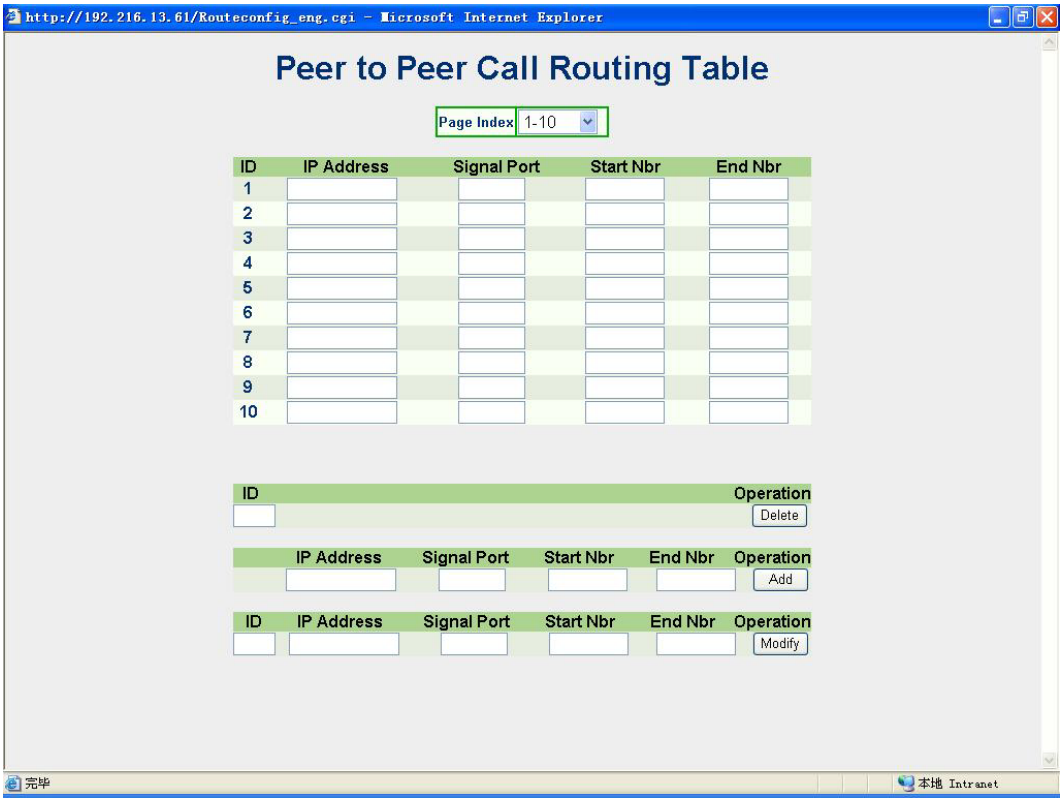


Figure 5-4 MG3000-R32 Peer to Peer Call Routing Cable Setup Interface

- Set up the IP address of the other peer in [IP Address].
- Fill in the call signal port number of the other peer in [Signal Port].
- Fill in the start number with the device port of the other peer in [Start Number].
- Fill in the end number with the device port of the other peer in [End Number].

Operation Notes: any direct modification to the contents of the table will be regarded as invalid unless it is performed with the buttons of [Delete], [Add] and [Modify] below.

5.6 IP/PSTN Routing and Number Transfer Table

In the 【Quick Setup】page click【IP/PSTN Routing and Number Transfer Table】button go to new page as the figure 5-5 shown：

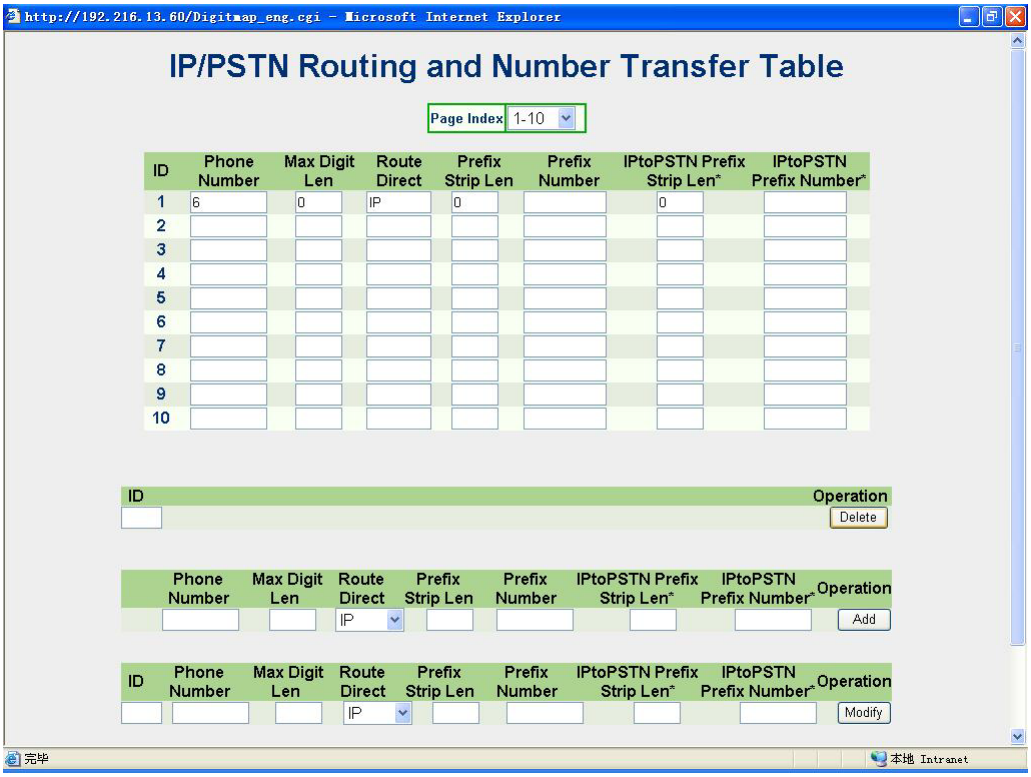


Figure 5-5 MG3000-R32 IP/PSTN routing and number switching interface

Surface indication:

- 【Dial-out number】: The N digit number in front of prefix callee number.
- 【Matching length】: The caller number length of dial-out.
- 【Routing direction】: The routing of caller, IP/PSTN.
- 【Prefix peel off digit】: For the dial-out number, delete the N digit from start to the end
- 【Prefix number】: Add the prefix number in front of dial-out number.
- 【IPtoPSTN, prefix peel off digit】: If the routing is in IP situation, when the IP network has the trouble, the signal would be indirection to PSTN, thus the callee number could be able to switching, the callee number is deleted N digit number from start to the end.
- 【IPtoPSTN, prefix number】: If the routing is in IP situation, when the IP network has the trouble, the signal would be indirection to PSTN, thus the callee number could be able to switching, the callee number is added as the number string.

The principle of IP/PSTN routing and number switching list:

- The prefix digit number is match to the dial-out number, and the length is complete

matching, then the route is succeeded.

- If there are many office directions and the lengths are matched, then the maximal length of office direction is setting as the standard. The figure 5.5 is illustrated if the user dial out the phone number 01062303100, then switching the callee routing and number switching according to the second route.
- IF the matching length is 0, then it is no need matching the length, and 【dial-out number】could be matched successful.
- If the【prefix peel off digit 】and【prefix number】could be configured in the same time, the principle of number switching is deleted firstly, and then add the new prefix. It is illustrated as the second router of the figure 5.5. If dial 01062303100, then firstly delete the 3 digits, it become 62303100, then add new digit 0 in the prefix, finally the dial-out number is 062303100.

The configuring operation explanation: To operating, just modify the content of list directory is invalid, it is especially to performed by the【delete】,【add】and 【modify】button.

For example, request all the domestic call pass the IP and all the city call are passed the PSTN. Due to the initial number of domestic call is 0, so in the figure 5.5, all the ID office direction is configured in 1. If there is the especial number as the 010 initially and dial through PSTN, then it could be configured as the second office direction of ID, meanwhile switching the number. The rest, initial number from 1-9 would be routed to PSTN, which is ignored since the default is routed to PSTN.

5.7 Configuration save and restart

After the configuration is finished, the next step is save the configuration and restart the PC. Click the 【save & restart】 button on the 【quick configuration】page, the equipment would be save the configuration automatically and restart the PC. After the restarting, the telephone connects the FXS and could able to using.

6 Basic configuration

6.1 Setting the network parameter

The network parameter could be configured on the【quick configuration】page, and also could be configured on the 【Network Configuration】, it is illustrated as the figure 6-1:

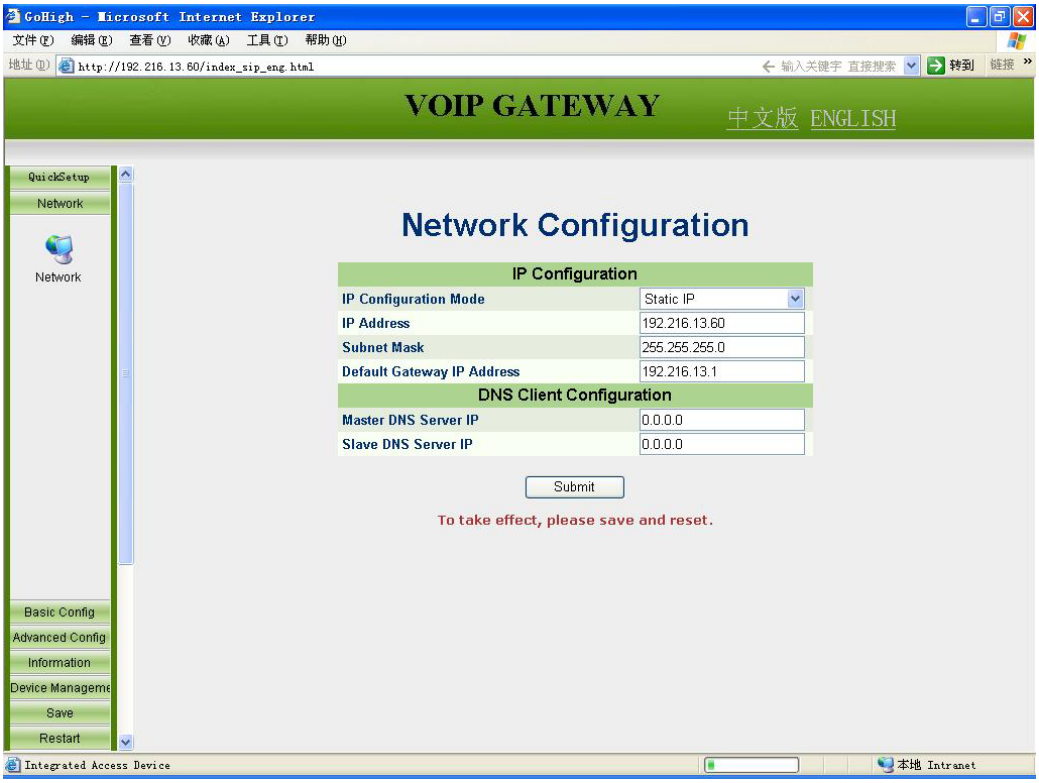


Figure 6-1 MG3000-R32 the network configuration page

The operating details please read 5.1section.

6.2 Setting the port number

Please ensure the MG3000-R32 software version before the configuring the port number. H.323 version please read the 6.2.1 section; SIP version please read the 6.2.2 section.

6.2.1 The basic port configuration of H.323 version

In the section it is introduced the basic information in the 【basic configuration】-> 【port configuration】, and the advanced configuration would be introduced successively on the extend chapter according to the function.

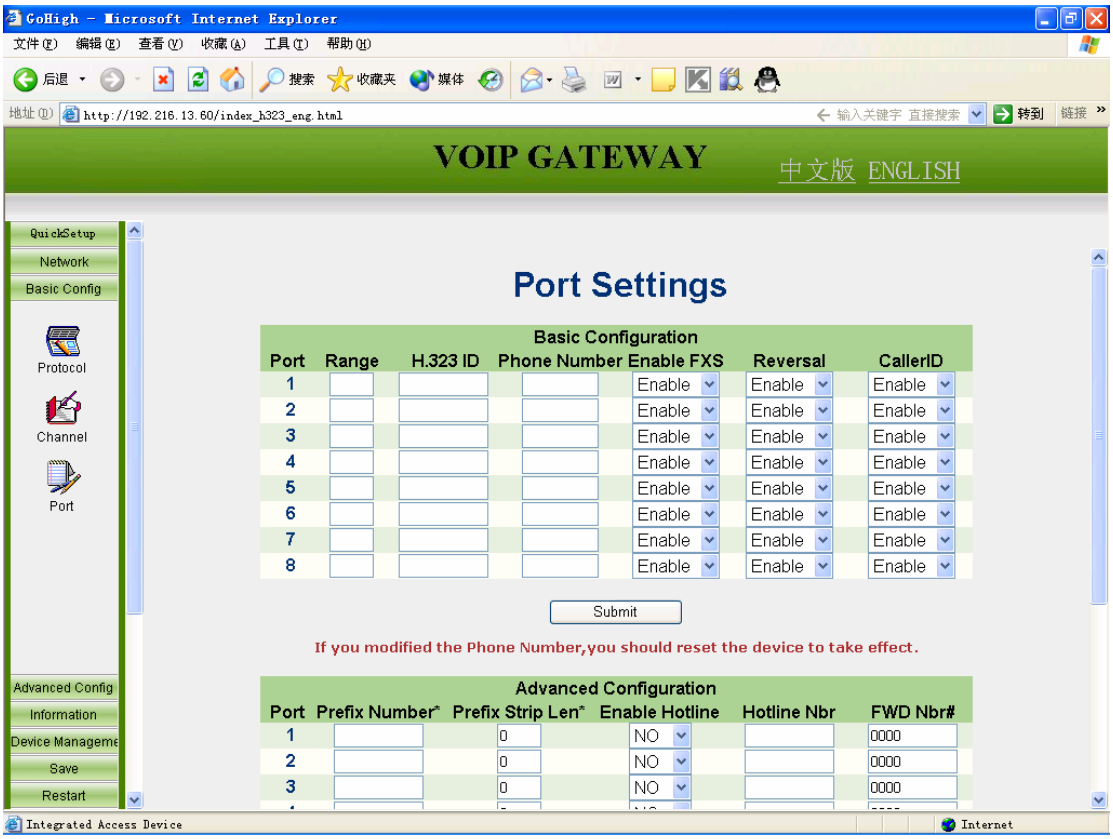


Figure 6.2 Port setting on the MG3000-R32 in H.323 version

The operation details please read 5.4.1 section

6.2.2 The basic port configuration of SIP version

In the section it is introduced the basic information in the 【basic configuration】->【port configuration】, and the advanced configuration would be introduced successively on the extend chapter according to the function.

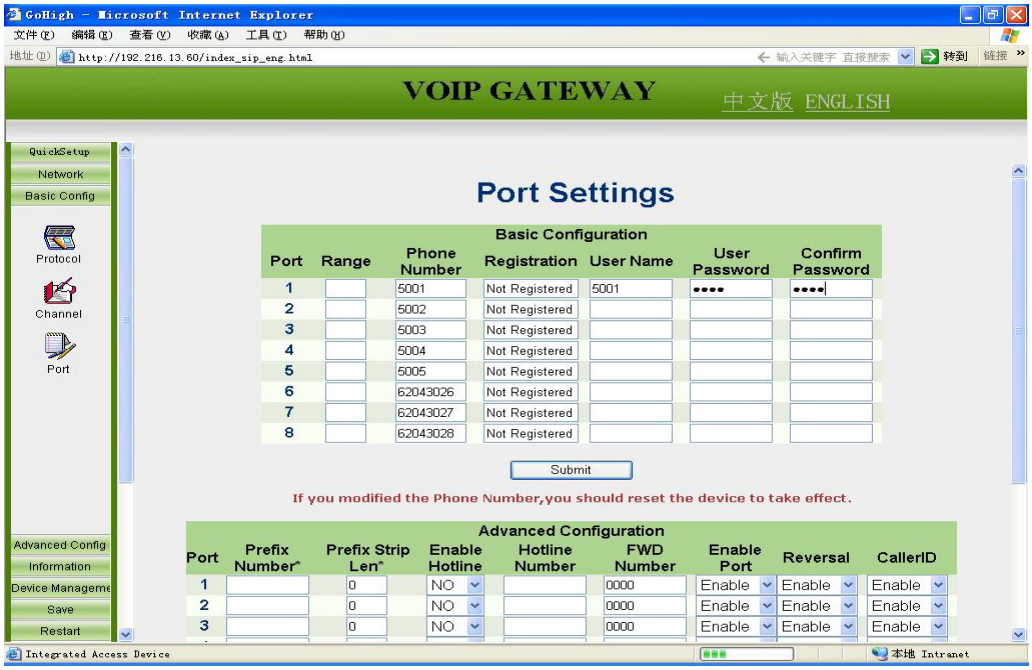


Figure 6.3 Port setting on the MG3000-R32 in SIP version

The operation details please read 5.4.2 section.

6.3 Configuring the peer to peer route

The point to point could be configured except in 【quick configuration】-> 【peer to peer routing table】page, it is also configured on 【advanced configuration】->【peer to peer call routing table】, it is showed as the figure 6-4:

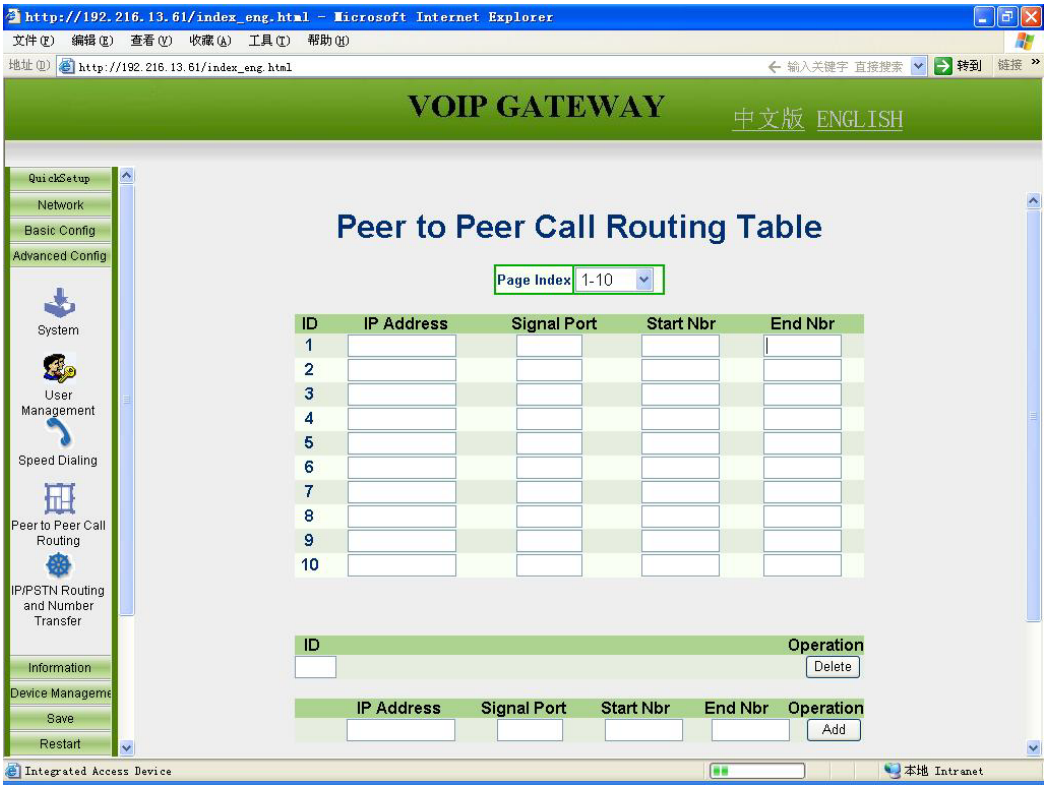


Figure 6-4 the interface of peer-to-peer call routing table

The operation details please read 5.5 Section.

6.4 IP/PSTN Routing and Number transfer

IP/PSTN routing choice and number switching could be configured except in 【quick configuration】->【IP/PSTN routing and number transfer table】page, it is also configured on 【advanced configuration】->【IP/PSTN routing and number transfer table】, it is showed as the figure 6-5:

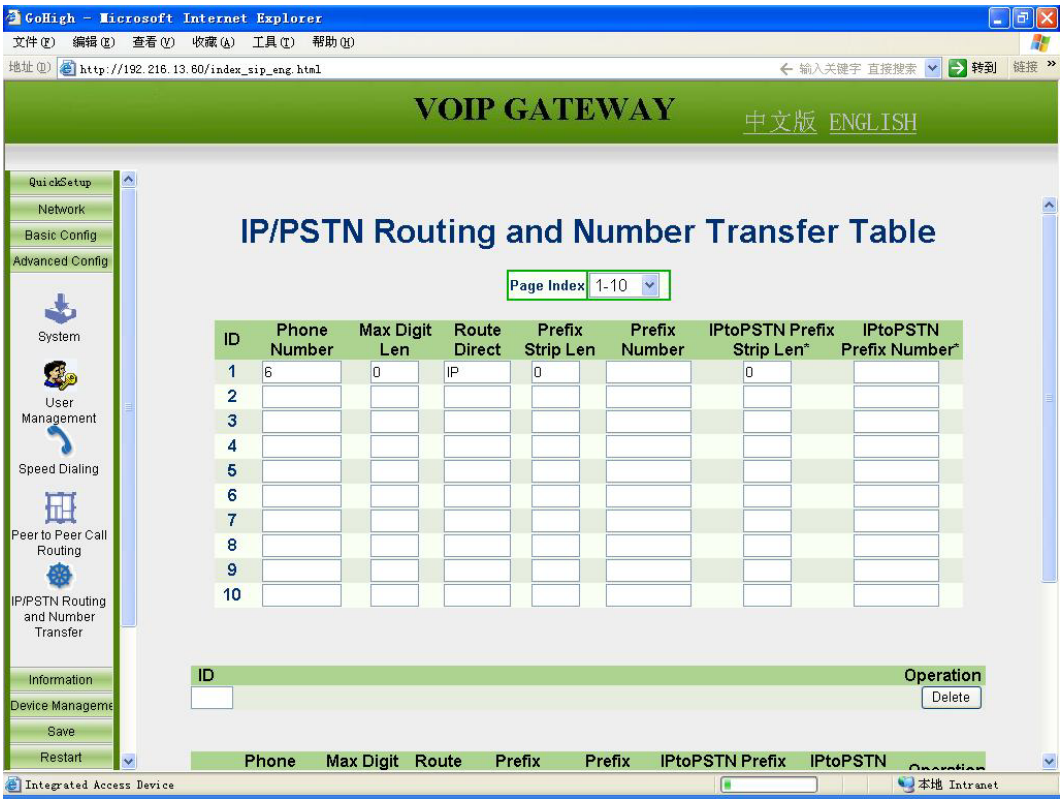


Figure 6-5 the interface of IP/PSTN routing and number transfer table in MG3000-R32

The operation details please read 5.6 Section.

6.5 Speed Dialing

The user could recorder the common telephone number by speed dialing table, as long as dial the speed key and then the system could mapping it as the original number to call out. For example, fill in the 123 in the 【speed number】, and fill in the 12345678 in the 【original number】, then when the user dial the 123 digit number, the system will automatically dial 12345678.

To configuring the speed number in the 【advanced configuration】->【speed dialing table】, which is illustrated as the figure 6-6:

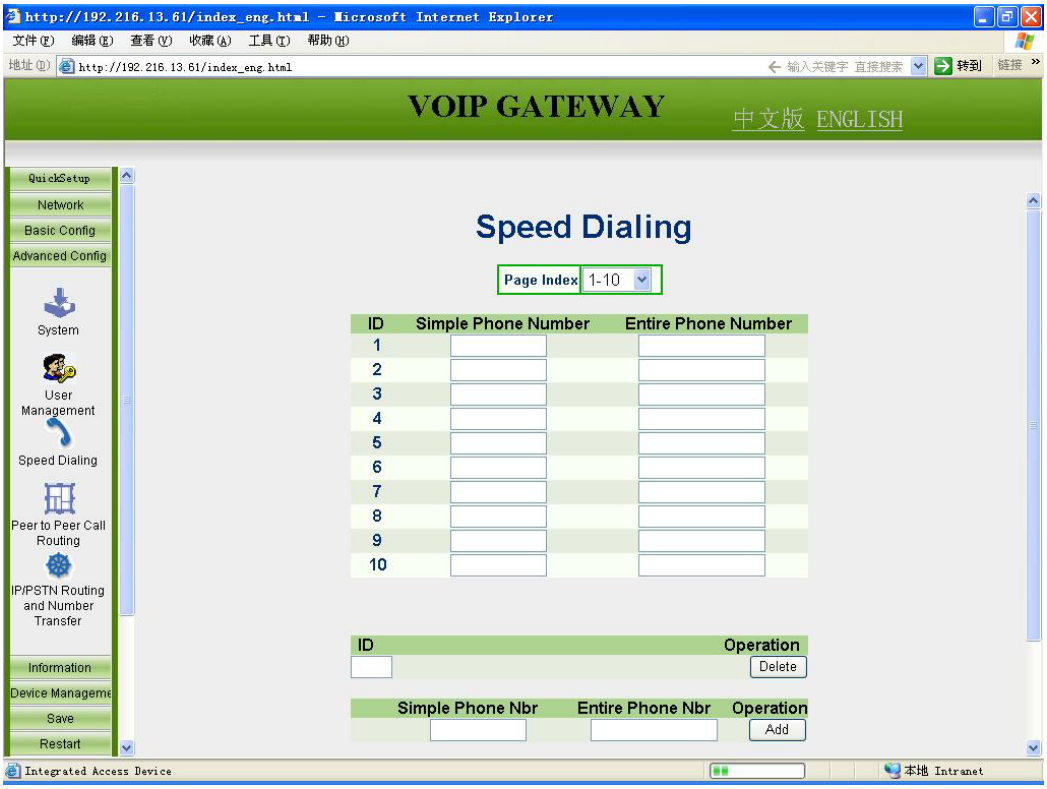


Figure 6-6 the interface of speed dialing in MG3000-R32

When the speed number will be deleted, just fill in the ID in the first rows, and click the delete button.

When the speed number will be added, just fill in the ID and the original number in the second rows, and click the add button. The system would automatically add the new speed number in the top row.

When the speed number will be modified, just fill in the ID in the third rows, and add the modify content beside, finally click the add button.

7 The protocol parameter definition

Before configuring the protocol parameter please confirm the MG3000-R32 software version.

H.323version user please read the 7.1 sections; SIP version user please read the 7.2 sections.

7.1 H.323protocol parameter definition

The basic protocol parameter of H.323 is already introduced in 5.2.1section. In this section, the content is included other protocol parameter and call connection parameters, the encryption configuration would be introduced later.

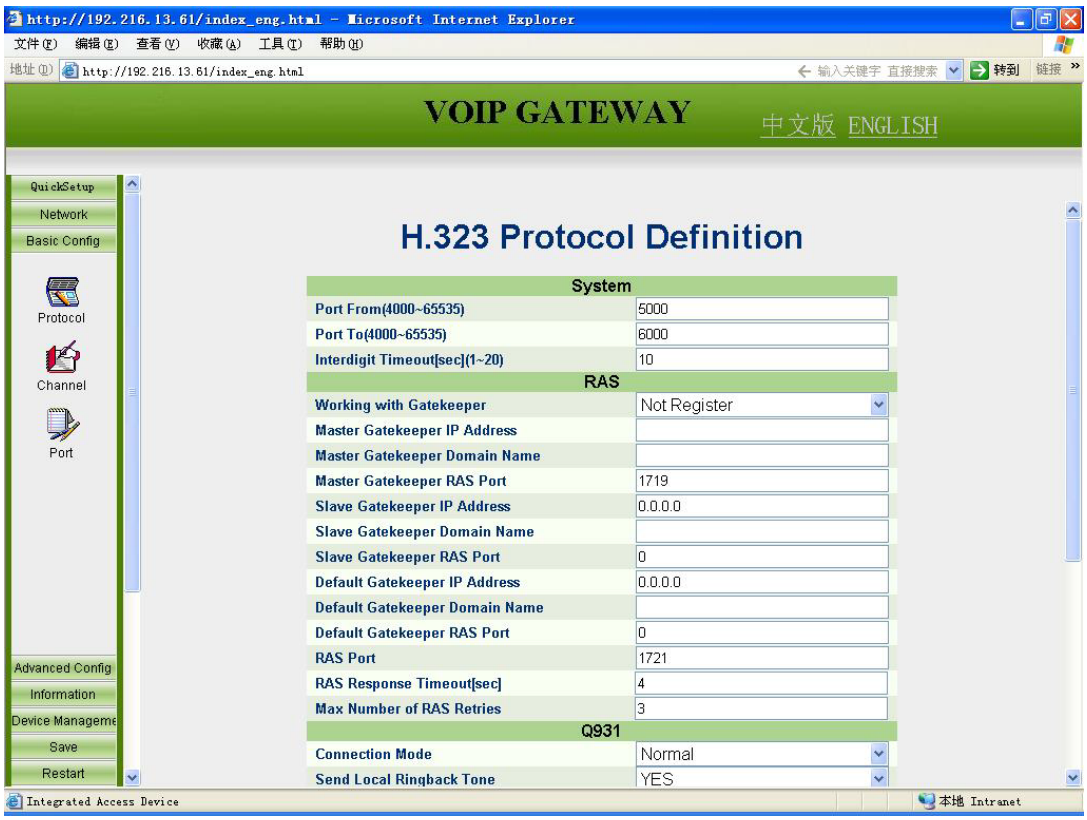


Figure 7-1 MG3000-R32 H.323 Protocol Definitions

The indication is showed:

7.1.1 Register parameter configuration:

- The details about【register on Gate Keeper】and【RAS port of default GK】please read 5.2.1 sections.
- 【RAS port】: Configuring the UDP port according to RAS protocol.
- 【RAS response overtime value】: Add the prefix number before the dial number.
- 【The maximal retest time of RAS 】: Configuring the maximal retest time of RAS

message.

7.1.2 Calling Related Parameter Configuration

- Beginning port number: defining the original beginning of the system port, from 4000~65535
- Ending port number: defining the ending of the system port, from 4000~65535. The ending port number should be at least 150 more than the beginning port number, in order to make sure there are enough ports available.
- Every dialing period overtime value: defining the overtime value of every dialing period. If more than the overtime value, the system will regard the user has finished dialing and send out the dialed number.
- Connect mode: set up the calling connect mode of MG3000-R.
- Sending local ring back tone: used for deciding whether to send local ring back tone. In the “linking mode”, when choosing “quick start” or “quick start + tunnel mode”, if you set up sending local ring back tone, the caller won’t hear the color tone of the callee and the media indicate tone of the callee exchanger. The caller could only hear the local ring back tone; if you set up not sending the local ring back tone, the caller can hear the color tone of the callee and the media indicate tone of the callee exchanger.
- Automatic answering: decide whether to start automatic answering.
- Overlap dialing: Activating the overlap dialing switch, the system will send phone number to callee by the overlap dialing manner. Your callee number will send every number one by one, or send together after dialing finish.(Waiting exceed the every dialing period or get the # number)
- Call signaling port: set up the transport layer port of the local gateway to receive Q931 information.
- Link overtime value: configure the overtime value when the calling terminal waiting for the callee terminal answering information
- Answering overtime value: configure the calling terminal send the “set up” information and wait for the ALERTING，CALL PROCEEDING，CONNECT，RELEASE COMPLETE or other overtime value.

- TCP connect Heartbeat value: In order to keeping the connection of TCP, the message of keeping the connection of TCP will be sent before the server cut this connection. TCP connect Heartbeat value means the interval value of sending configuration message to server, these value from 20~7200,unit is sec.

7.2 SIP Protocol Parameter Definition

SIP basic protocol parameter has been introduced in chapter 5.2.2. This chapter mainly introduces other protocol parameter and calling link related parameter. The encrypt config will be introduced in the following chapter. As the chart 7-2 below:

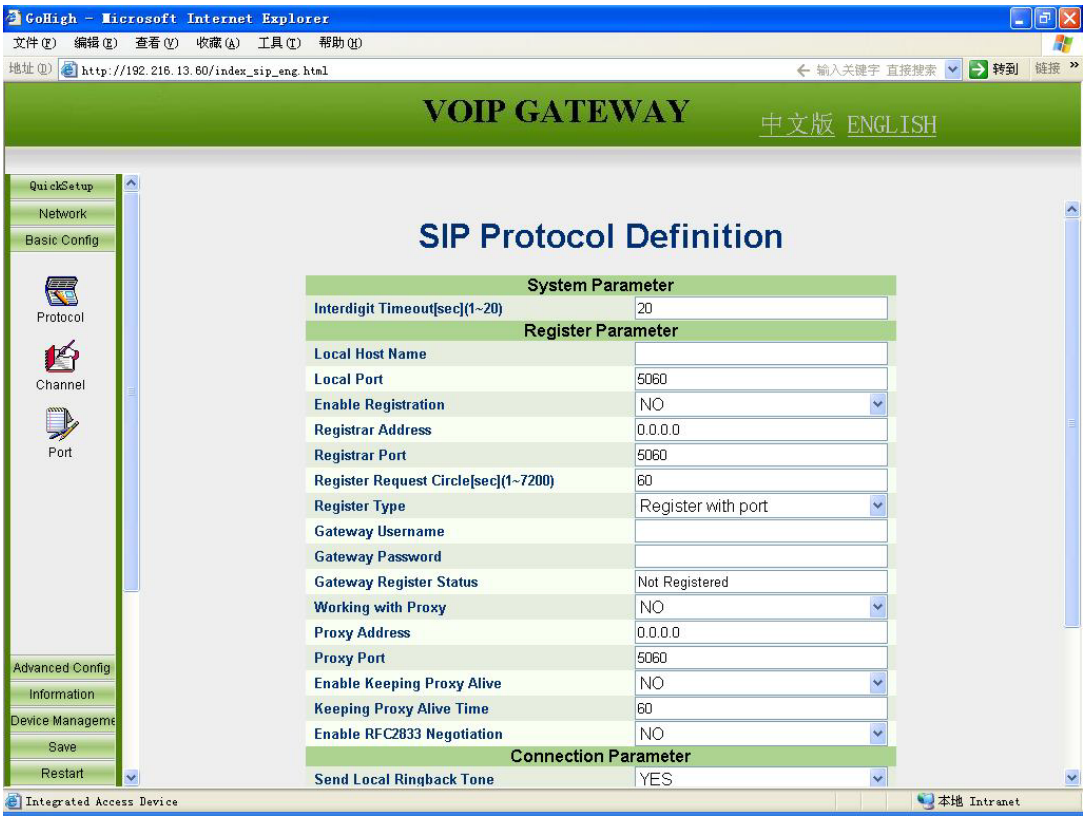


Figure 7-2 MG3000-R32 SIP parameters configuration interface

7.2.1 Register Parameter Configuration

- Local port: set up the UDP port numbers of the local computer SIP register information.
- Gateway register status: show the register status of the gateway register.
- Enable Keeping Proxy Alive: choose whether to enable keeping proxy alive. If choose “YES”, it will send information to the Proxy regularly, and shows the terminal is available.

- Keeping Proxy Alive Time: configure the interval time of sending information to the Proxy
- Enable RFC2833 Negotiation: when the DTMF mode is chosen as RFC 2833, choose whether carrying RFC2833 negotiation.

7.2.2 Calling Related Parameter Configuration

- Every dialing overtime value: defining the overtime value of every dialing. If more than the overtime value, the system will regard the dialing finished and send out the calling number.
- Sending local ring back tone: used for deciding whether to send local ring back tone. If you set up sending local ring back tone, the caller won't hear the color tone of the callee and the media indicate tone of the callee exchanger. The caller could only hear the local ring back tone; if you set up not sending the local ring back tone, the caller can hear the color tone of the callee and the media indicate tone of the callee exchanger.

7.3 Replacing Agreement Type

In the "Advanced Config->System Config" page, choose the signal protocol type, as chart7-3:

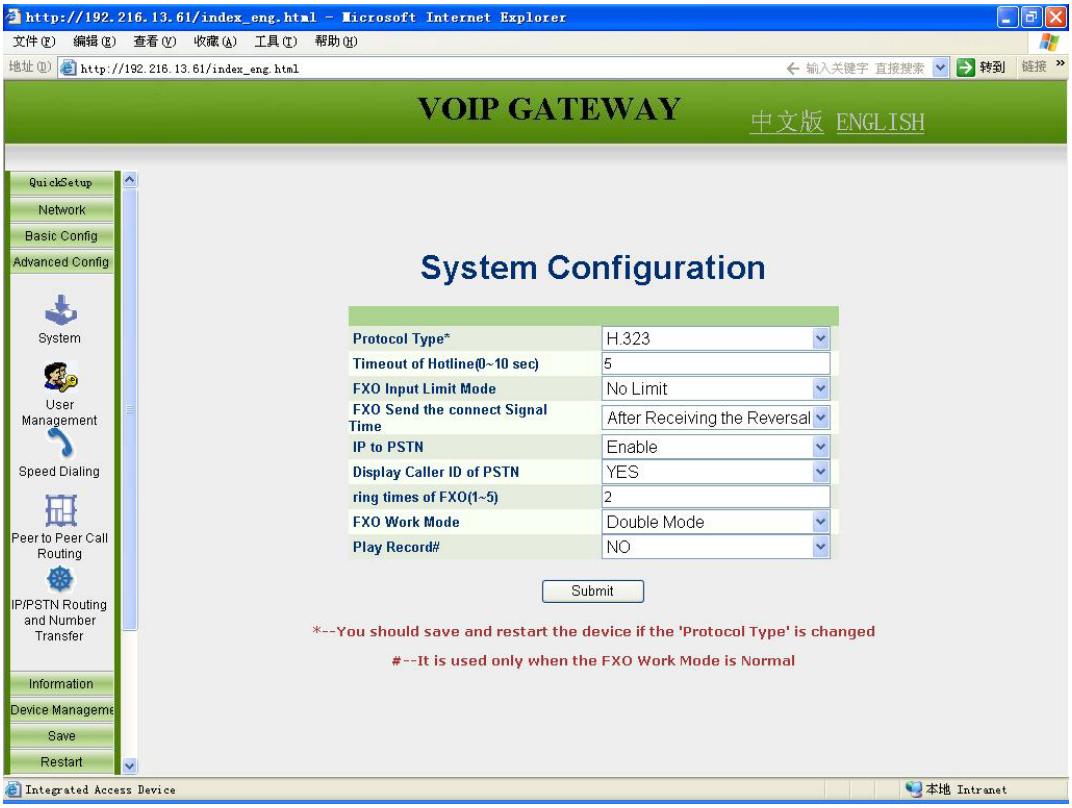


Figure 7-3 MG3000-R System Configuration interface

When switching between the SIP agreement and H.223 agreement, the set up has to be saved and restart the equipment, then it can take effect. Current controlled edition needs to eliminate the IE buffer memory.

8 DSP Detail Parameter Configuration

"DSP detail parameter set up" is in the "Basic Config-> Channel Setting" page, in order to set up the parameter used for building media channel, as chart 8-1:

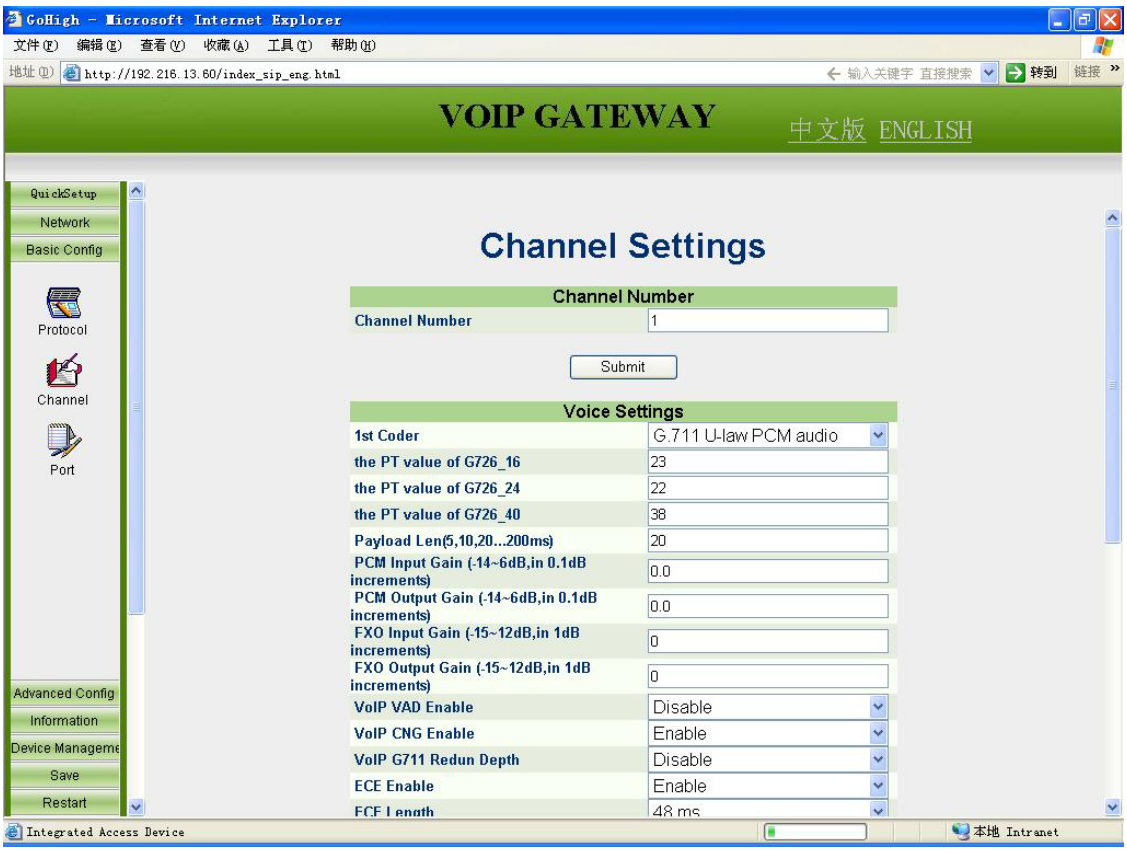


Figure 8-1 MG3000-R32 channel setting interface

Input the channel number in "Channel Number". If it is "0", it means all the channel will be configured the same parameter. When configure the voice channel parameter, it is suggested that the user use default parameter without modification, except "1st coder".

8.1 Voice Parameter Configuration

8.1.1 Configure Code Mode

- 1st coder mode: the first code mode of the channel, there are G.711 U-law PCM audio，G.723.1 audio，G.726_32 audio ，G.711 A-law PCM audio，G.729 A audio，G.726_16 audio，G.726_24 audio，G.726_40 audio availble.
- The PT value of G726_16: payload type of G726_16
- The PT value of G726_24: payload type of G726_24

- The PT value of G726_40: payload type of G726_40

8.1.2 Configure Voice Gain

- PCM Input Gain (-14~6dB, in 0.1dB increments): PCM input gain is used for zooming in received audio signal;
- PCM Output Gain (-14~6dB, in 0.1dB increments): PCM output gain is used for zooming in audio signal which has been sent out. The stronger of the audio signal sent out, the stronger of the anti-jamming ability during the signal transmitting.
- FXO Input Gain(-15~12dB, in 1dB increments): the Gain of FXO receiving audio signal.
- FXO Output Gain(-15~12dB, in 1dB increments): the Gain of FXO sending audio signal.

8.1.3 Eliminate Echo

- Eliminate echo: whether to support eliminate echo
- Eliminate echo length: define the trail wavelength of eliminating echo. The echo can be eliminated in the time length chosen by the user.

8.1.4 Voice Quality Parameter

- Packaging period (5,10,20…200ms): the period for packaging the voice data is the length of RTP package.
- Voice activity test: Whether activate voice activity test, whether send indicate package of mute, adopt which manner to send. This function can detect the voice transmission of the user. If there is no voice transmission, you can reduce the network burden through mute condense.
- Comfort noise: Whether generate comfort noise;
- G.711 redundancies deep: It means the redundancy deep of RTP when use the G.711 code mode. This function can forbid some voice uncontinuity cause by lost package in the network.

8.2 Fax Parameter Configuration

- Transmission mode: The Transmission mode of FAX, can be T.38 FAX Relay or penetrate (use the currently code mode of voice to code. Only can use penetrate

under the voice code manner of G.711)

- Fax maximum speed: The default value of maximum speed limited of T.38 FAX is no limited. When working by register manner, it is better to configure to Fax maximum speed of the server. When working by point-to-point manner, it is better to configure to Fax maximum speed of opposite server.
- ECM error correction: Whether activate T.38 FAX ECM error correction. The default value is yes.
- T.38 error correction system: The default value of T.38 fax error correction manner is redundancy.
- Fax data redundancy depth: T.38 FAX data redundancy depth can avoid to lost the package of FAX data. The default value is 0.
- T.30 message redundancy depth: T.30 message redundancy depth can avoid to lost the package of T.30 message. The default value is 3.
- T.38 fax gain(-15~0dB,ascending by1dB): The trunking ascending of FAX, the default value is 0.
- T.38 lost package hidden mode: The error hide mode of lost T38 Fax message. The default value is none.
- TCF mode: Training Check Field means there must have a TCF training course to ensure the going of circuit before send message, the default value is distant end.

8.3 DTMF Parameter Configuration

- DTMF Transmission mode: The transmission mode of DTMF message must activate echo elimination when second dial.
- RFC2833 gain load type (96~127): RFC2833 gain load type of DTMF.
- RFC2833 redundancy gain load type (96-127): RFC2833 redundancy gain load type of DTMF message(the value of RFC2833 gain load type must be different from RFC2833 redundancy gain load type)
- RFC2833 transmission mode: Setting RFC2833 transmission mode.

9 FXO Analog Junction Function Configuration

9.1 Configure Normal Mode of FXO

The work mode configuration of FXO is in the "Advanced Configuration->System Configuration" page, as chart 9-1:

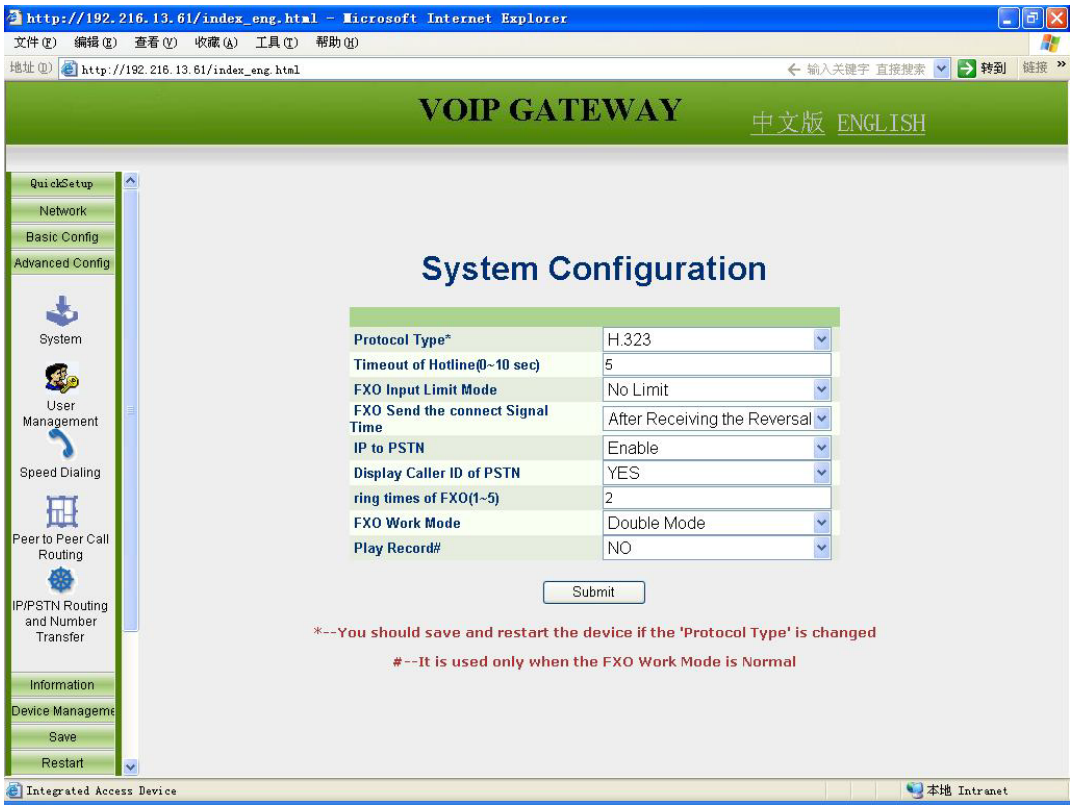


Figure 9-1 MG3000-R32 System Configuration

It can be defied as "normal mode", "Attendant console mode" and "Double Mode". Only all cards is 4FXS4FXO card can be defined as "Double Mode".

When FXO working under "normal mode", the call of PSTN pass through out PSTN to FXO can listen voice cue or traditional second dial voice. When you put called number to dial IP phone, you must set the correct IP/PSTN routing to callee IP side phone. When FXO working under point-to-point mode, you need to set point-to-point call routing, the detail configurations are in the Figure 5.5 and Figure 5.6.

When you make an IP call from IP network to FXO, you can use primary dial to make FXO auto regenerate number (The configuration of FXO number regenerate see as Figure 9.4) or dial the number of FXO. You make second dial after you hear the dial tone, it can dial the number of

PSTN by the outline of PSTN connecting the FXO.

The item of “play voice recording” is valid only under normal mode. If you choose “yes”, the call comes from PSTN will send second dial indicate tone, such as “welcome to Datang Gohigh, please dial extension number, if you want to get number infromating you can dial 0”; If you choose “No”, the call comes from PSTN will send second dial tone.

9.2 Configuration ““Double Mode” of FXO

When the FXO is working under Attendant console mode, you must set the attendant console number of FXO. The incoming call from FSTN to FXO will be auto go to homologous attendant console number. The configurations of attendant console number are in the “Port Configuration” of “Basic Configuration”. See at figure 9-2. You must set the correct IP/PSTN routing to attendant console number. When FXO working under point-to-point mode, you need to set point-to-point call routing, the detail configurations are in the Figure 5.5 and Figure 5.6.

The mode of IP phone making a call from FXO to PSTN is same configuration with normal mode.

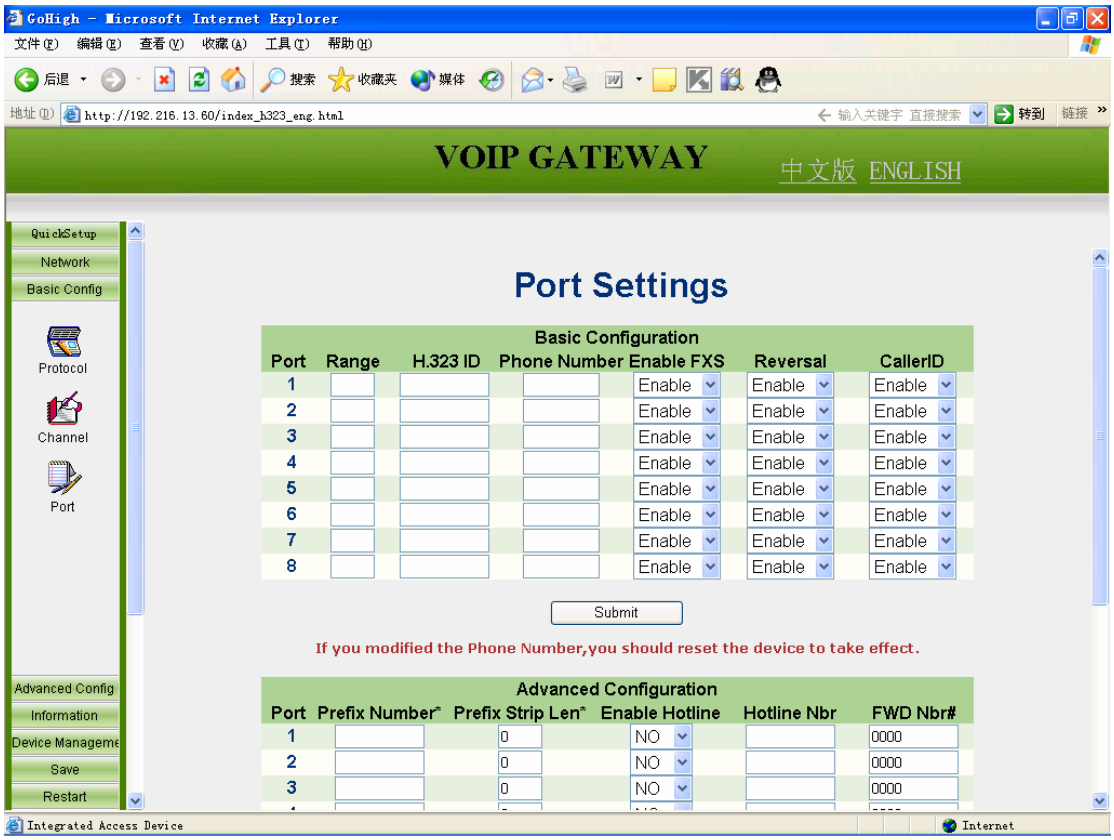


Figure 9-2 MG3000-R32 Port Settings

9.3 Configuration "Double Mode" of FXO

The equipment with 4FXS and 4FXO can be set to "Double Mode"

When FXO working under "Attendant console mode", the port 1 correspond to port 8, the port 2 correspond to port 7, the port 3 correspond to port 6, the port 4 correspond to port 5.

When a call from the port of FXS to the number of PSTN, the call will outgoing from the FXO corresponding with his FXS.

When a call send to the outline PSTN of FXO, the telephone of FXS corresponding to FXO will ringing. The FXO port is used by FXS only in the same equipment.

9.4 FXO number regeneration

If it should realize the one dial to the FXO port, and make the second number redial to the PSTN through the FXO port, and it must call the callee number beside IP and based on the FXO port. For example, one of the equipment in the configure of the telephone number base on the FXO port is 010, the dialing number beside IP is 01062303100，and make this calling to pointing the equipment, and the FXO port will redial the number 01062303100.

If you want to transform the number and redial，please transform the number in the interface of 【Basic configuration】->【port configuration 】, as figure 9-2 is showed. For example, "prefix " configure 3，"prefix number " is "0"， and delete three unit number in front of 01062303100， and get the number of 62303100, then add the prefix number "0", and get 062303100, FXO make the 062303100 redial .

9.5 PSTN calling authority

The function of white list is aimed at the application of PSTN for incoming call from the FXO port, it controlled the caller number beside the PSTN, and the PSTN Caller number in the white list can permit incoming call, and it does not permit if it is not in the white list. For the incoming call can restrict or not, it is according to the PSTN caller number, so it should open the FXO to test PSTN and display the function, the detailed configuration is detailed in 9.6 section.

In the interface of 【advanced configuration 】-> 【system configuration 】, can configure the "FXO pattern of call restriction" as the figure 9-1 is illustrated.

If you choose “unrestricted”; all the PSTN caller number will permit to incoming.

If you choose “restricted the non-user, continue to sending the ring back tone ”. And the caller from PSTN can make the incoming call from FXO port out of white list. The caller from PSTN can incoming call from the FXO port, and you can listen to the continued ring back tone, but the FXO port does not pick up and cannot establish this calling.

If restrict the non-user, “pick up for 1 second and hang up”, and the caller from PSTN in the white list can incoming call from FXO port; out of the white list, the caller from PSTN. Incoming call from the FXO, pick up for the O port and the caller will be appear the busy tone.

If it will make a restriction of PSTN caller number, make the permitted & incoming caller number from PSTN to input in the 【advanced configuration 】,【user management of the white list】as the figure 9-3 is illustrated:

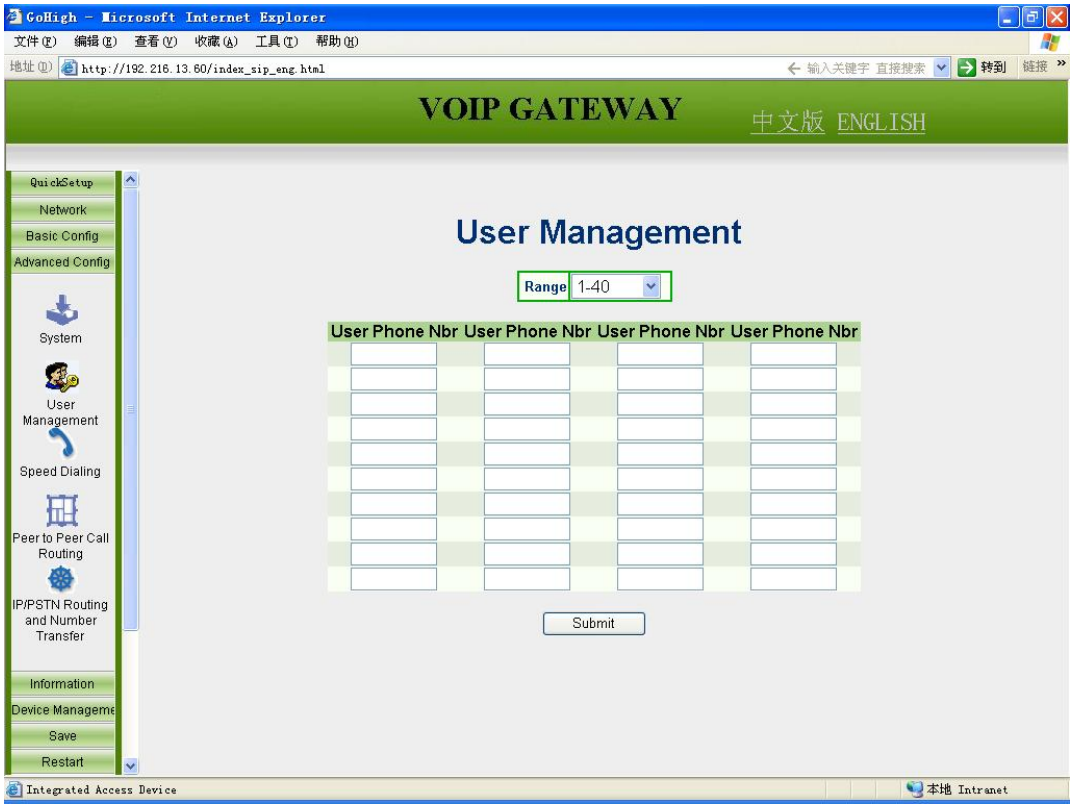


Figure 9-3 MG3000-R32 the interface of user management in white list

9.6 PSTN incoming call list

FXO receive the call from PSTN, and through the second dial (common mode) to dial the extension, or call it automatically to the attendant number（attendant mode），or call it directly to the fixed FXS port（double module mode），if you want to the extension of callee display the caller

number from PSTN, you should configure as follows:

- Callee extension be set to active caller ID;
- The extension of callee used the caller ID telephone;
- In the page layout of 【advanced configuration 】-> 【system configuration】, it will "display the caller number from PSTN", and to configure "yes" display PSTN, figure 9-1 as follows;
- In the page of the【advanced configuration】-> 【system configuration】, it will configure the "times of shaking bel by FXO port is 1~5" to more or up to the amount of 2, figure 9-1 as show;
- The exterior line of PSTN connects with FXO port and realizes the function of caller ID.

Attention: after register the platform, the platform demand that it should authorize the caller number, maybe the number change to be the caller number from PSTN and result in un-authorization, thus it should not realize the function of caller number from PSTN.

9.7 The signal advice of charge for charging by PSTN

For the calling from IP to FXO, and through the FXO port and redial the number to the PSTN, the billing signal inform can be choose: when the FXO is picked up and begin to billing; 2. The callee from PSTN is picked up and begins to billing.

In the page layout of 【advanced configuration】-> 【system configuration】 to configure the option of the "FXO send the connect time", figure 9-1 is as followed. If you want to get the demand when the FXO is picked up and begin to billing, you should choose "sending in normal"; if the callee from PSTN is picked up and begins to billing, you choose "after receiving the reverse signal "

10 Management

10.1 look over the port estate

In the page layout of【Information】-> 【port】, you can check the style and state of each port. In the edition of H.323, you can check the register estate for the equipment. And the page will be refurbished automatically in each 5senconds, and the figure 10-1 is followed.

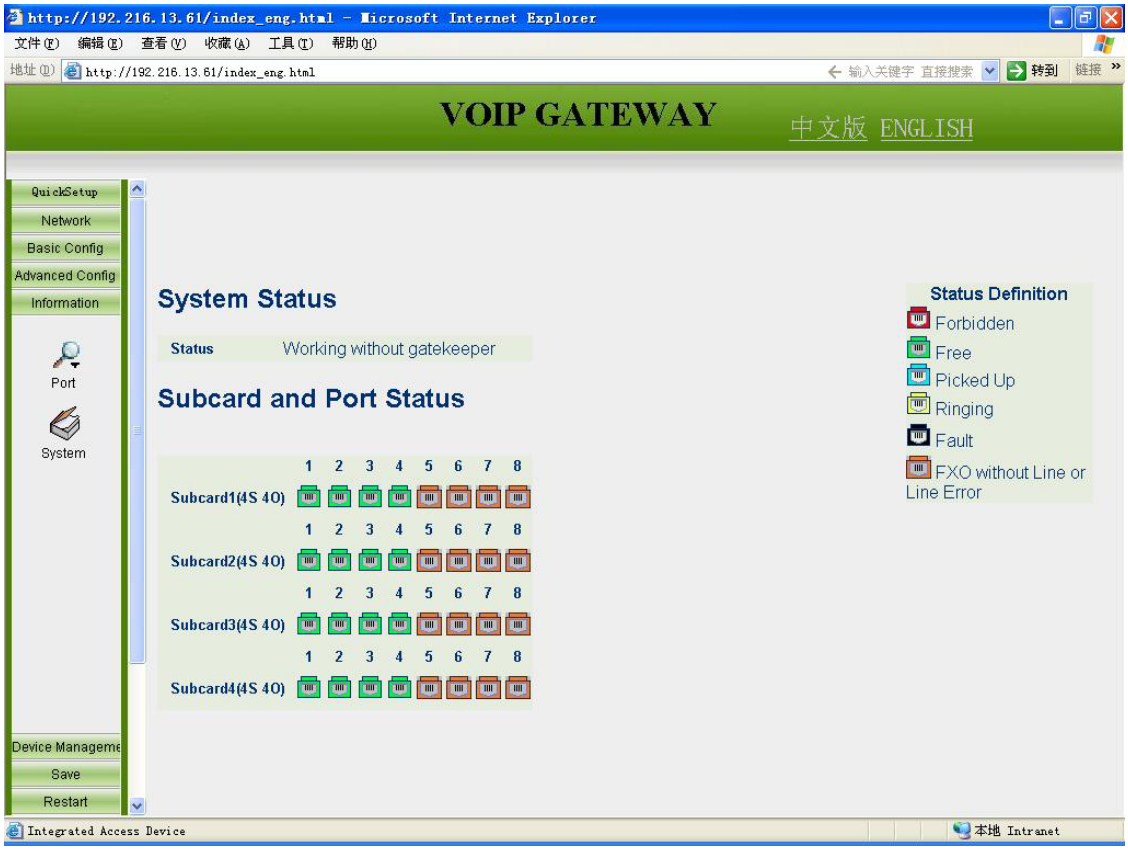


Figure 10-1 MG3000-R32 interface of port estate

10.2 Check the version information

In the page layout of 【information】-> 【system information 】, you can check the edition information of the hardware and software, and the page layout can refurbish automatically for each 10 seconds, the figure 10-2 is followed:

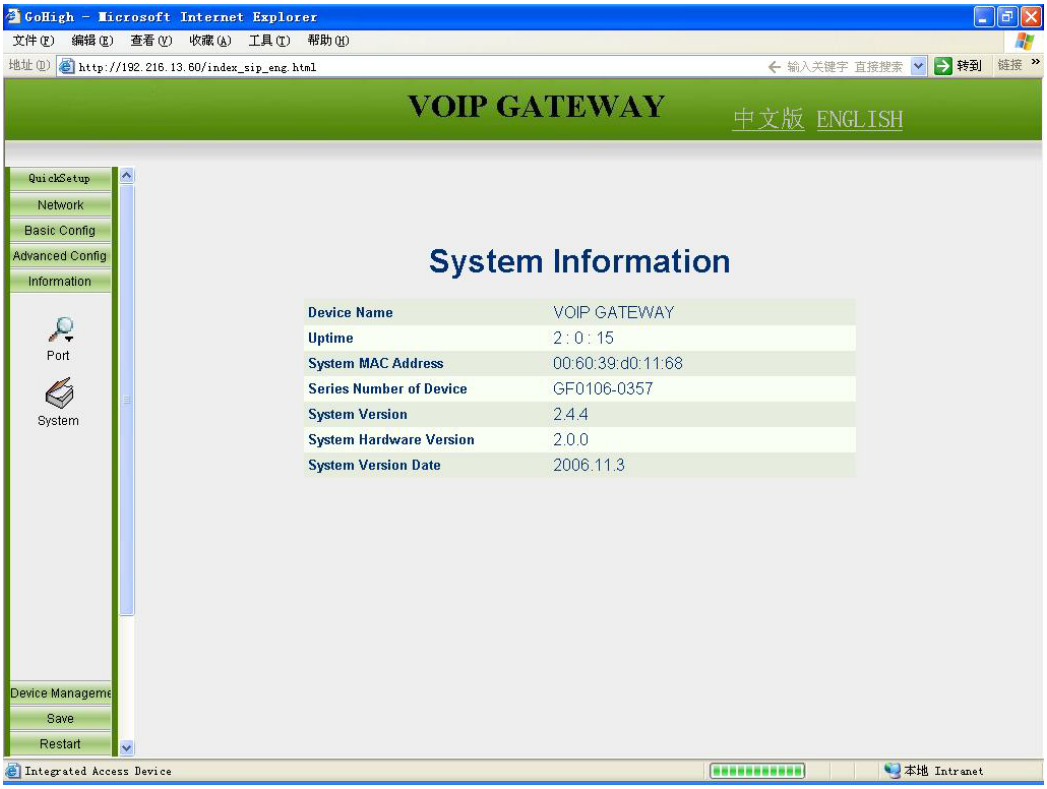


Figure 10-2 MG3000-R32 interface of system information

10.3 Firmware upgrade

In the page layout of 【equipment management】-> 【upgrade management】 and can make the edition upgrade.，down load of configuration document or operation of picking up configuration document, the figure 10-3, first choose the transmission protocol, and can choose the HTTP（recommendation）or TFTP protocol.

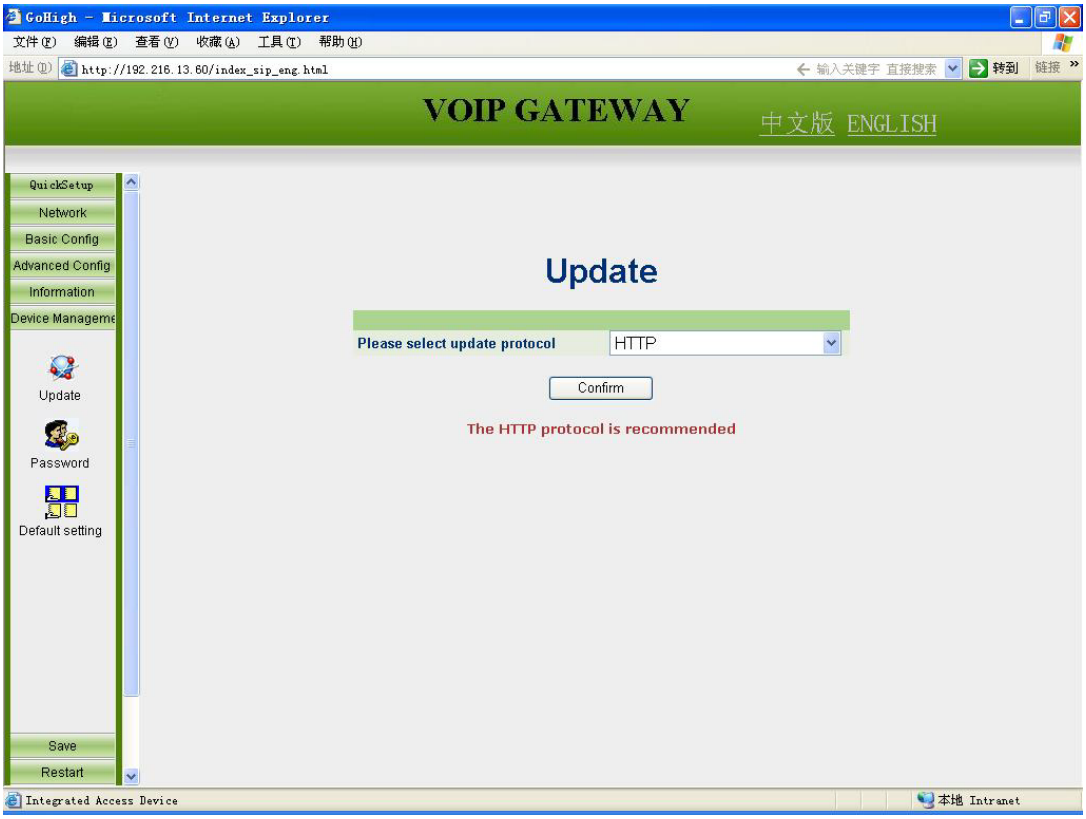


Figure 10-3 the interface of MG3000-R32 upgrade management

If you choose the protocol of HTTP protocol, click “yes” button and enter into the interface of upgrade edition, figure 10-4 as follows;

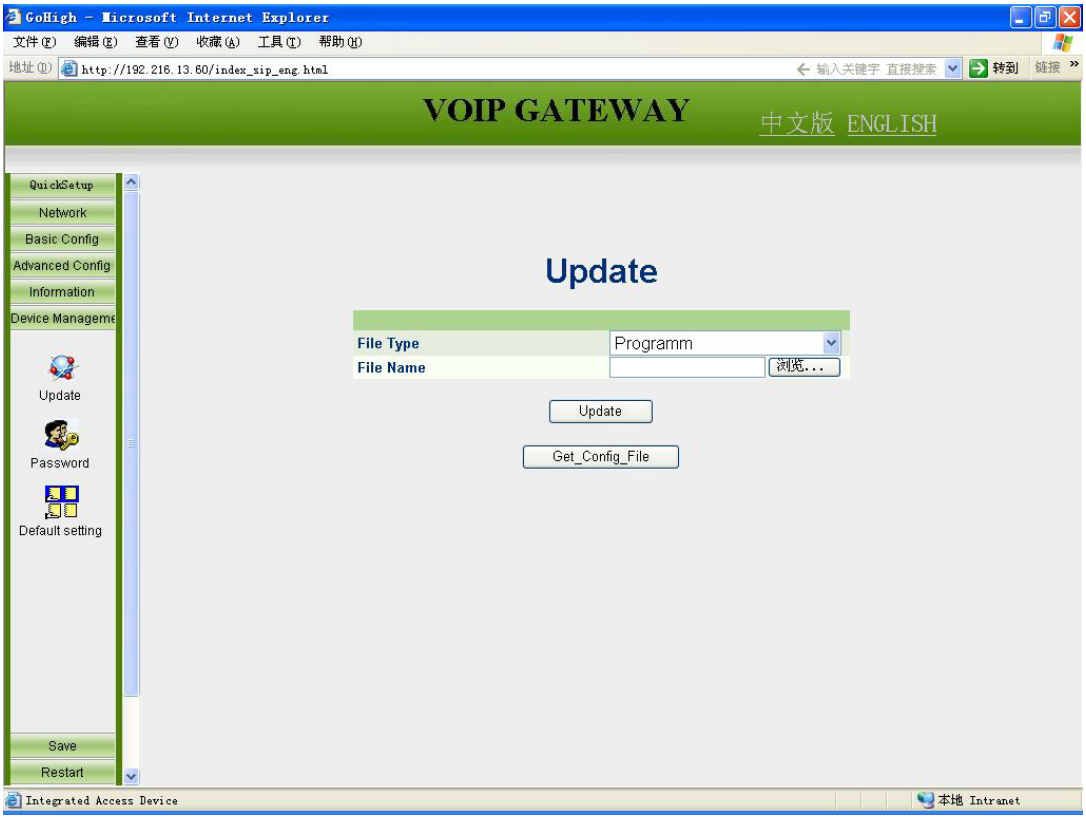


Figure10-4The interface of MG3000-R32 HTTP upgrade management

- If need the upgrade program, In the “document type” choose “program document”, click the button of “browse” to choose the program document which will download, and click the button of “upgrade”, after finishing the upgrade, the system will prompt the “program document download successes, it need reboot the equipment can go to in effect”.
- If need upgrade the configuration document, and in the “document type ” choose “configuration document”, and its operation is the same as program document and attention, there is two configuration document for H.323 protocol: the system configuration document (mg3000_r.ini) and the protocol configuration document (mg3000r.val); For the SIP protocol，it only has system configuration document (mg3000_r.ini). When upgrade the configuration document, the file name should be mg3000_r.ini or mg3000r.val.
- LOGO is a photograph of company and the product name is showed in the top of the web, it can upload. And the size of the LOGO should be 224（width）×86（length），logo should be in the center of the file name. The product name and picture for up loading should be 654（width）×86（length）, the center of file name should conclude name.
- If need take the configuration information, click the “taking configuration document”, and save the configuration document to this computer.

If choose the TFTP protocol，click “yes” button and into the interface of upgrade edition. The figure 10-5 is showed as the following:

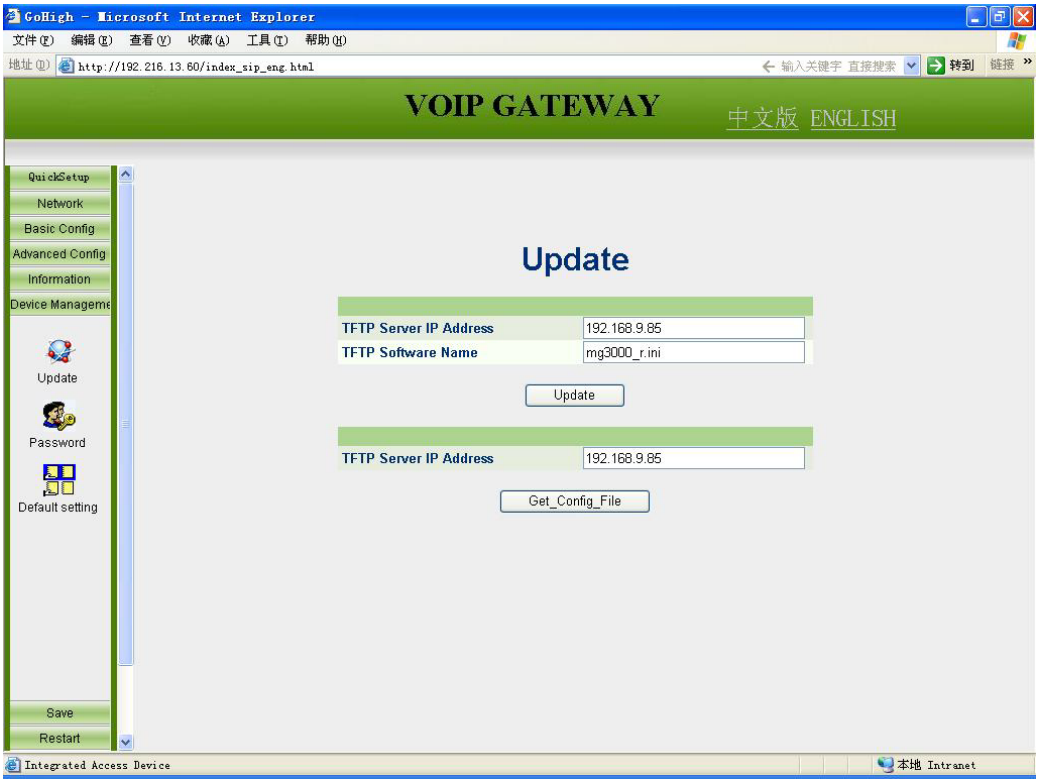


Figure 10-5 MG3000-R32 TFTP interface of upgrading management

When use the TFTP protocol to upgrade, it needs to provide the TFTP sever and the IP address which is download and upgrade the document.

Attention: when configuring the document of mg3000_r.ini and the mg3000r.val, their configuration document and the file name couldn't be changed, (among which, the configuration document mg3000r.val just use for H.323 protocol edition). The other document will be made for a program document to download. Before upgrading the program, we should running the TFTP Server in the TFTP server, and click the button of "upgrade" to make it. If taking the current configuration information of the equipment, click the button of "take the configuration document".

10.4 Amending the user password in web

In the page layout of 【equipment management】-> 【resume the equipment】, it can amend the WEB console password, the figure 10-6 is as follows.

Attention: it amending the user password, and the user name cannot revise.

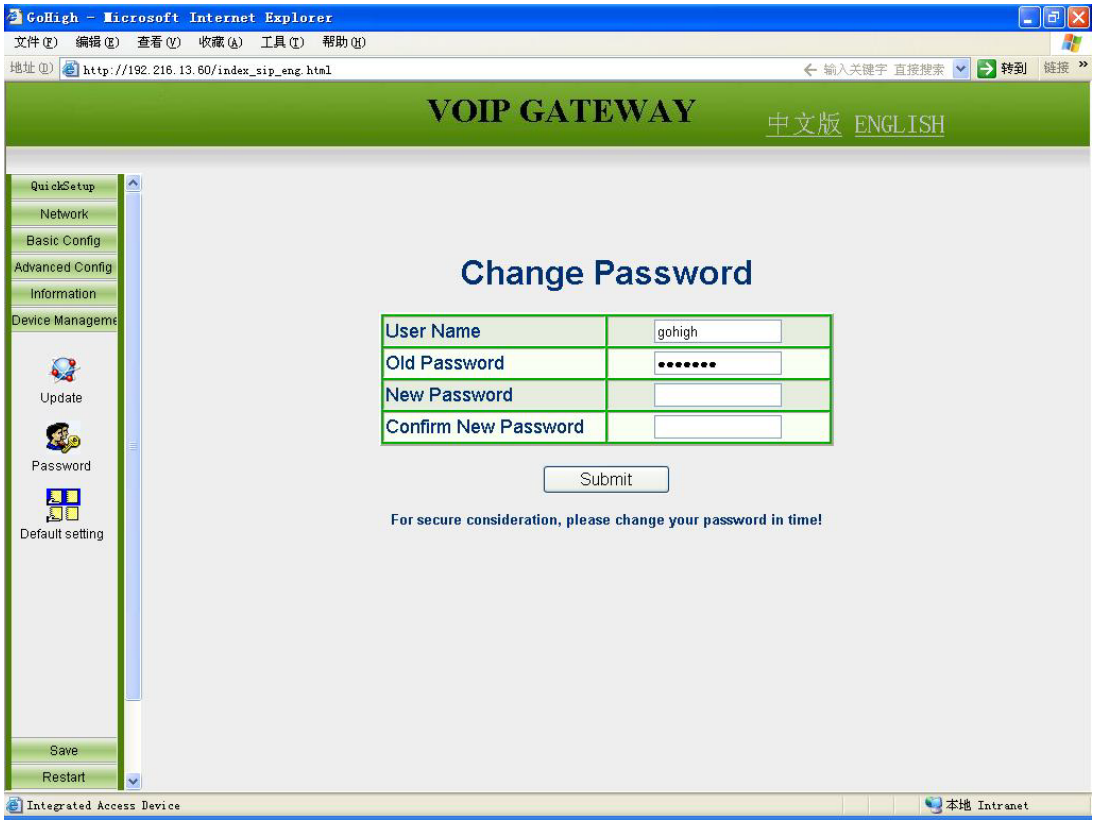


Figure 10-6 MG3000-R32 interface of passport management

10.5 Resume the beginning configuration

In the page layout of【equipment management】->【resume the configuration 】, it can make it to be acquiescence, the figure 10-7 is showed as follows:

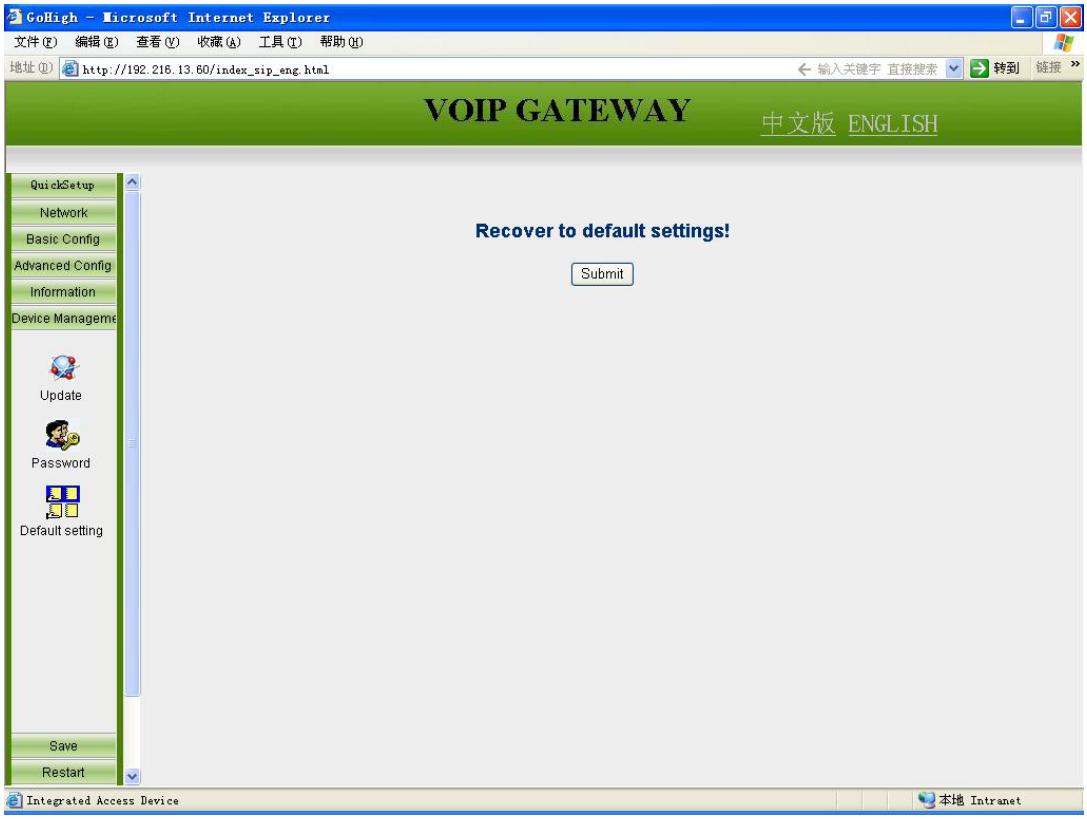


Figure 10-7 MG3000-R32 the interface of resume the configuration

Click the “yes” button, and after resume the acquiescence setting, it should be save reboot and it will become effective.

10.6 Save Configration

The “save config” page will save configuration as the figure 10-8.

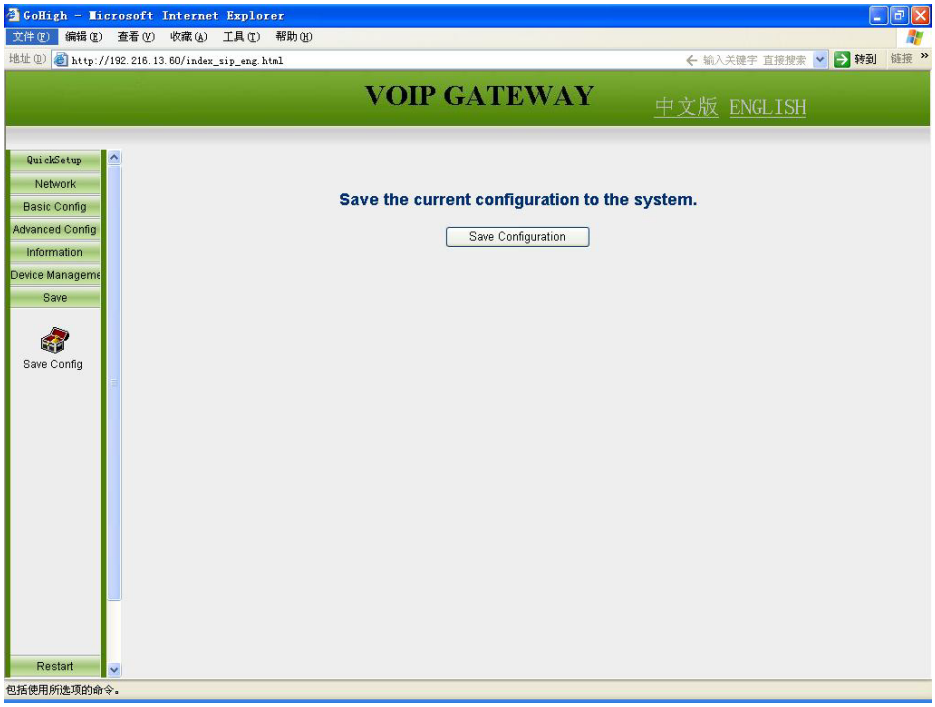


Figure 10-8 MG3000-R32 Save config page

10.7 Restart Device

The 【Restart Device】page will have the MG3000-R restart, as the figure 10-9 shown.

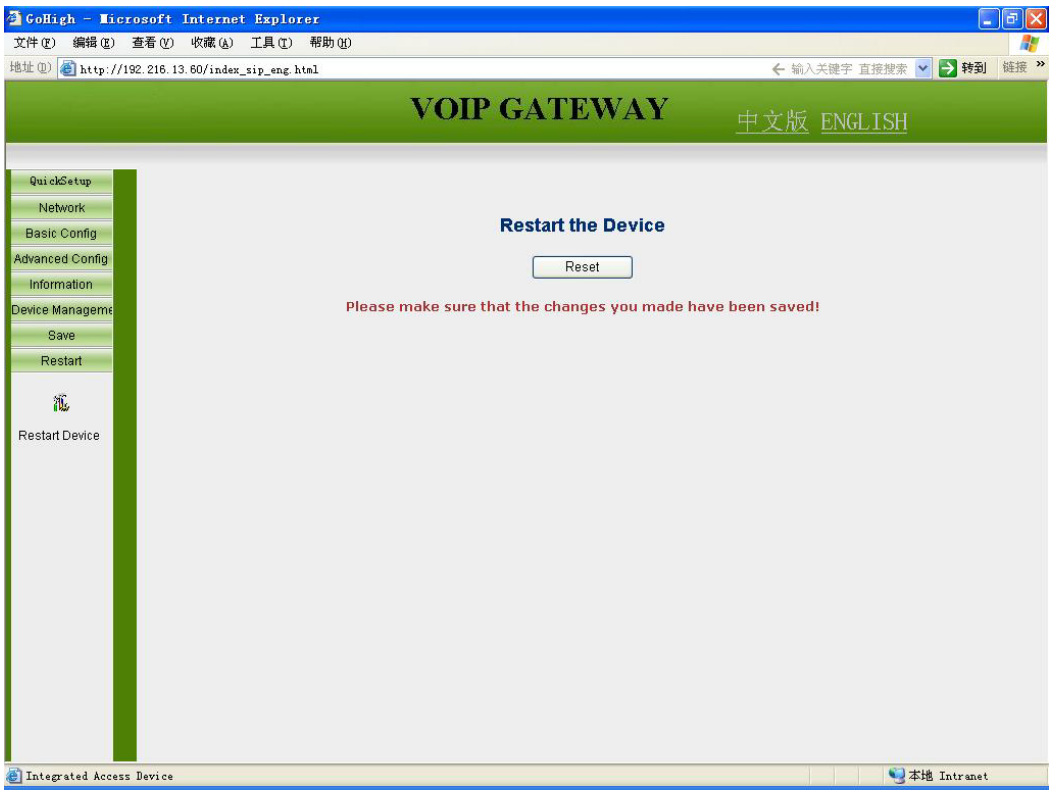


Figure 10-9 MG3000-R32 Restart

11 Other configration

11.1 FXS port send Caller ID

In the page 【Basic config】-> 【Port settings】page, there is "CallerID" listbox in the Basic configuration. Choose "enable" to active it. As the Figure 6-2 or 6-3 shown.

11.2 FXS Send polarity reversal signal

In the page 【Basic config】-> 【Port settings】page, there is "CallerID" listbox in the Basic configuration. Choose "enable" to active it. As the Figure 6-2 or 6-3 shown.

11.3 Hot Line Setting

In the【Advanced Config】-> 【System configuration】page "Timeout of Hotline(0-10 sec", as the figure 9-1 shown.

If you want to open an port's hotline function, please set the 【Enable Hotline】 on the page 【Basic Config】-> 【Port Settings】 to "Yes". Input the hotline number , as Figure 6-2 or Figure 6-3 shown.

If the port opened the hotline function, after the port pick up, if the end user didn't dial any number in the time out time, the gateway will dial the hotline number automatically.

11.4 Encrption

In the 【Basic Config】->【Protocol】page, select Encryption Type and key under your service provider's advise.

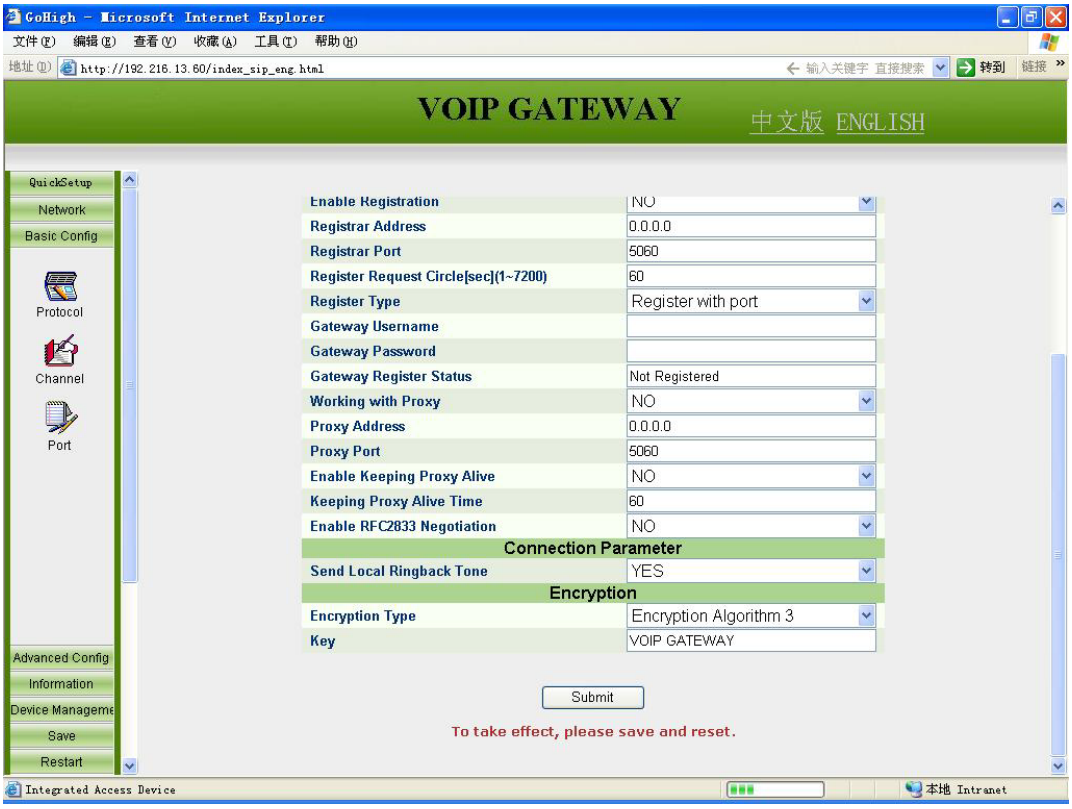


Figure 11-1 MG3000-R32 Protocol parameters page

In the item 【Encryption Type】, there are options including no encryption, encryption algorithm 1, encryption algorithm 2, encryption algorithm 3. In the item 【key】 ,input your key.

12 FAQ

12.1 Difference of the 3 work model

	Normal Mode	Attendant Mode	Double Mode
Call from PSTN to FXO port, the callee is on the VOIP side.	Caller can use two stage dial method to make calls like this gateway's voip account.	When calls from PSTN into VOIP, the callee is fixed as a voip number. When calls arrived, the pstn call will be transfer to a VOIP number automatically.	The callee should be the FXS port only.
VOIP users call PSTN by one stage dial or two stage dial.	VOIP user can call the FXO gateway's VOIP number freely.	Same to Normal Model	In this mode, PSTN side can't receive call from VOIP. The PSTN side can receive call from FXS side.

12.2 Three use case of number transfer

12.2.1 The item "Prefix Strip Len" and "Prefix Number" of 【IP/PSTN routing and Number transfer Table】

It can be used in:

1、 Local FXS port telephone start a call. The number transfer will take place when call to VOIP or PSTN numbers.

2、 In normal mode or attendant mode, calls from PSTN arrived the CO line that FXO port connected, the number transfer will take place on the second stage dial number(the normal mode) or the attendant number (the attendant mode).

12.2.2 The item "IPtoPSTN Prefix Strip Len" and "IPtoPSTN Prefix Number" of 【IP/PSTN routing and Number transfer Table】

This type of number transferr will take place only in one case. It must fit such condition:

1、 This call's route is belong to IP;

2、 The IP route failure backup function is open.

3、 This call will be reroute into the pstn side for some reason such as IP access broken.

For example:

One user in Beijing are using MG3000-R 4S4O subcard. The FXO port connect one PSTN line with Beijing number. The FXS port Registered into the SoftSwith and can make call to Shanghai by call termination.

This user set the number prefixed 021 route to VOIP, and have the IP route failure backup function. The IPtoPSTN Prefix Strip Len set to 0, IPtoPSTN Prefix Number set to "17910*1234567890*123456". The ""17910*1234567890" is the access number and card number, "123456" is the password. For example, if user try to call 021-12345678, for the ip network error, call will back to PSTN. The gateway will dial 17910 by PSTN firstly. After received the IVR, it will dial card number, after received the IVR again, it will dial password. It will dial 02112345678 by PSTN at the end. In an other way, there is a one stage dial IP card number that is 96446, you could set the IPtoPSTN Prefix Number set to "96446". When the PSTN backup take place, the device will dial "9644602112345678" by PSTN at one time.

12.2.3 The Prefix number and Prefix Strip Len in the page of 【Port Settings】->【Advance Settings】

This number transferr take place in Normal mode or Attendant mode only. The Call from VOIP to PSTN by one stage dial. The FXO will regenerate the number again.

For example:

One fxo port of the MG3000-R gateway's VOIP number is 19. The Prefix strip Length is setting to 2. The Prefix number is 0. The FXO port will connect to PSTN.

The VOIP side user call to 191860, This call will route to a MG3000-R32. The gateway will cut the prefixed 2 number 19 of "191860". The result of number is 1860. The gateway will send this number out by PSTN.

If it need the region number 021, it can be added prefix 021, the gateway will add the prefix 021 after delete the two number. The result callee number is 0211860.

12.3 Number regeneration Usage

The FXO port work as the analogue trunking used in IP to PSTN terminateon. The end user don't like to hear the dial tone two times. It will need the gateway to have user dialed number regenerated. In this case, it need the softswith's support.

In h.323 protocol, MG3000-R32 can support Prefix register. The Gatekeeper must support prefix Register too. And it must forward the calls which have such prefix into MG3000-R32 gateway. After the gateway received calls, it can regenerate the used dialed PSTN number again and dial it on the PSTN line. another method is the gateway have a trunking route setting to this gateway. All caller have the same prefix should route to this gateway.

In the SIP protocol, the softswitch must support trunking route mode. It must have the FXO gateway's prefix number set to the gateway.

12.4 How to larger the voice plus

In the 【Basic Config】-> 【Channel Setting】page, The "PCM Input Gain" and "PCM output Gain" value (as Figure 8-1 shown) will adapt the voice plus, to make a better volumn. If the calls cross the FXO port, The "FXO Input Gain" and "FXO output Gain" item will adapt the volumn too. But the FXO port plus will affect the signal lose and large the noise. So the value should between –5db~+5db.

12.5 Why is the call from VOIP into FXO gateway busy

Please confirm the FXO gateway's voip number is correct and the caller have the right to call to this number. After that, please check if the gateway's work mode is not under double mode, for the FXO gateway can't receive call from VOIP under double mode.

12.6 Why it is no answer after the FXS or FXO dialed

If the call establish after dialing finished longly, it is for there is no # as a finished number. The device must wait timeout to send out number. So the call is delayed. The solution is dial a "#" after finished dialing. The callee number will be send out right now.

If the end user is not used to dail "#" additionally, it could have the 【number internal timeout】value in the 【SIP Protocol Definition】 smaller.

12.7 Why the second dial failed in the call from PSTN?

If the second dial failed in normal mode or connect attendant failed under attendant mode in the call from PSTN, please check the IP/PSTN Routing and Number Transfer Table. If the second dialed number or attendant number have the right route?

12.8 How to check the device's serial number and Mac address.

Please look for it at 【Information】-> 【System Information】page, as the Figure 10-2 shown.

12.9 Why the web page can't visited after change protocol.

In the version before V2.4.4, it must clear the IE cache after change protocol. The detail as follow:

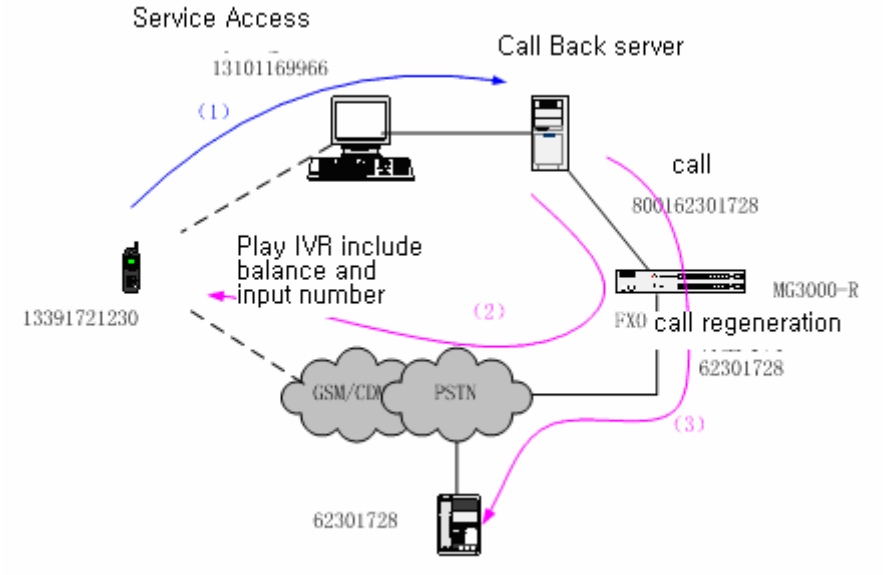
1、 After change protocol, save and reset the device;

2、 In the internet explorer, select "Tools"=>"Internet Options", click "Delete files", on the pop up dialogue box , don't select "Delete all offline content", click 'ok'.

3、 Login into the web page again.

13 Application example

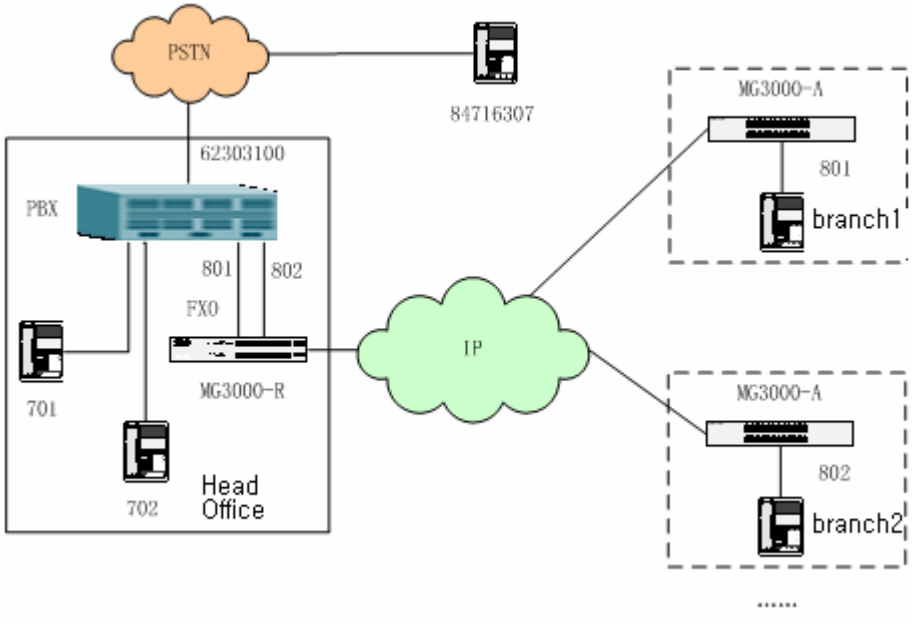
13.1 Call back



The time of FXO gateway send time: it must set to “after received polarity reversal signal”. Attention: It is conflict with the non number regeneration application. In this case , it can’t set to “after received polarity reversal signal”. For the FXO port don’t regenerate phone number, the FXO gateway will never receive the polarity signal , so it will never send out connect message. The caller will hear the ringback tone only before timeout.

no Register mode: When MG3000-R used in Call back for terminateon, it can be used in no Register mode. The call will route to this gateway directly.

13.2 Remote extension



Setting:

Remote side use MG3000-A device:

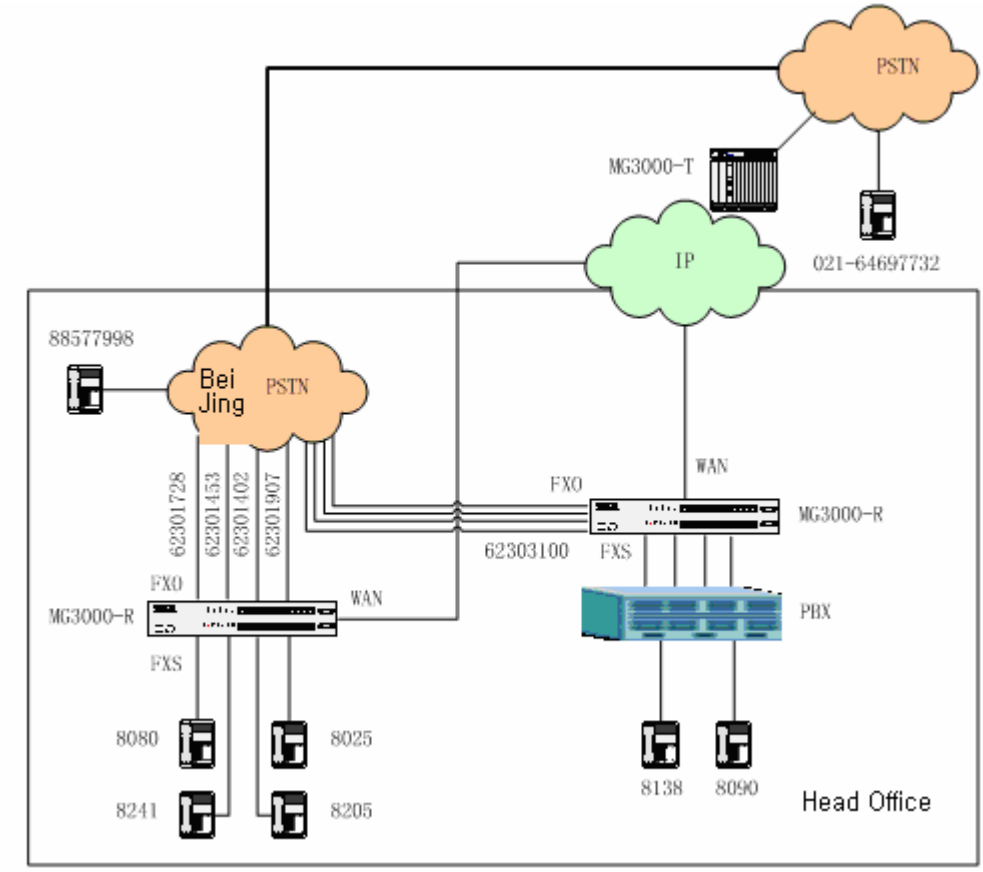
- Active that port's hot line function, and set the time to 0 second. The hotline number set to the MG3000-R's VOIP number.
- The extension number set to the remote PSTN number.

The MG3000-R in beijing:

- Have the FXO gateway 's voip number set to 91728;
- Set to attandent mode. The attendant number is the MG3000-A's extention number.

Addtionally, those two device should set the same DTMF method.

13.3 Intelligent Route application.



Parameter setting:

- FXO gateway mode: double mode.
- IP/PSTN route table: set the call out route rule;

FXO port ringback time: if there is Caller ID display from PSTN, this parameter must >1.